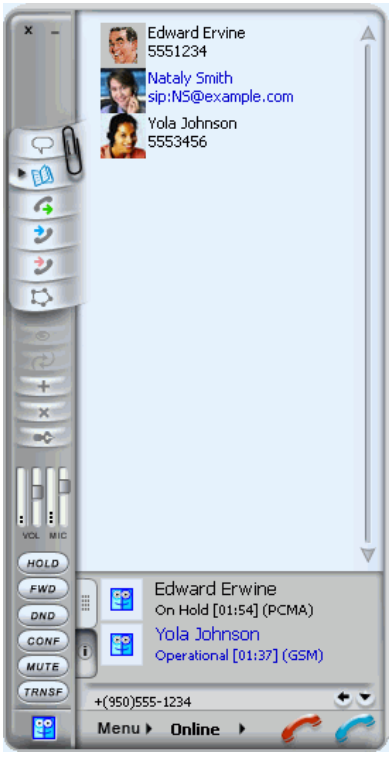
Version 1.65

SJphone™

Windows XP version



User's Guide

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I. Welcome to SJphone™

SJphone™ is a softphone that allows you to speak over Internet with any desktops, notebooks, PDAs, stand-alone IP-phones and even with any conventional landline or mobile phones. It supports both SIP and H.323 industry standards, and is fully inter-operable with most major Internet Telephony Service Providers (ITSP) and VoIP software and hardware.

For calling over the world you need to open an account with an IP-Telephony service and to get their service profile, or to make their **SJphone™** profile yourself.

Advanced users may build their own Voice over IP networks using either an H.323 Gatekeeper, SIP proxy, IP-PBX, and other components. They can also connect to conventional telephones using H.323/SIP Gateways (such as Cisco 26xx/36xx/37xx/53xx).

1.1. SJphone™ Features

SIP Compatibility

allows you to speak over Internet

- directly with most other SIP-softphones and stand-alone SIP-phones (such as Cisco 7940)
- with other people in your company through your SIP IP-PBX system or Proxies (such as Asterisk, Broadsoft and others)
- with all other people around the world using most SIP ITSP accounts such as FWD (**www.fwd.pulver.com**)
- with a regular telephone using most SIP Gateways

SIP features:

- TCP support
- TLS tunneling support
- Full support for SRV/NAPTR records and transport layer enhancements
- Concealed Caller ID
- DNS support in the SIP stack

H.323 Compatibility allows you to speak over Internet

- directly with any other H.323-softphones, and stand-alone H.323-phones.
- with other people in your company through your H.323 IP-PBX system, such as Cisco Call Manager
- with all other people around the world through any H.323 ITSP accounts
- with a computer or telephone on a private network through a **Gatekeeper**
- a regular telephone using most **H.323 Gateways**

Instant Messaging:

- Jabber: Presence
- Jabber: Simple Chat
- Jabber: vCard Display/Editor
- Jabber: Customizable Rules for Placing Calls from the Roster
- Jabber: full support for TLS transport

Features that make SJphone™ Easy to Use:

- Skins allow you to change the appearance of the **SJphone™ Main** panel
- Call any host by its name or IP address
- Proven Interoperability with most IT service providers (you can download profiles for some of them)
- Fast Service switching between ITSP, IP-PBX, Gatekeeper, Gateway, or SIP Proxy settings and GUI

- Contacts, call logs, and speed dialing
- Outlook and Opera contacts
- Call a computer or telephone on a private network through a **Gatekeeper**
- Call transfer, hold, and mute
- Can automatically accept incoming calls, or reject them immediately (the "**Do Not Disturb**" mode) or after a specified time
- Automatic discovery of neighbor **SJphones™**
- Respondent list allows you to maintain and manage multiple calls
- Dial pad to type in numbers and send DTMF signals, sent using the either as audio, INFO message, or out-of-band methods.
- Audio Wizard that allows the user to set audio options easily and quickly.
- Selectable Playback/Recording/Ringer devices switched on-the-fly
- Input and output sound level indication and control
- Automatically adjustable microphone volume
- Automatically adjustable silence detection level
- Support for most USB phones (several drivers are downloadable)
- Support for most popular codecs: G711, GSM, iLBC, (G729 available only in commercial versions)
- Support for wideband codecs
- Noise reduction
- Automatic Gain Control for microphone (software/hardware mode)
- Automatic Echo Cancelling (AEC)
- Automatic NAT detection
- Calls from/to a private IP-network using the NAT traversal technology
- Voicemail Message waiting indication
- Vertical Service Codes

Features for Advanced Users:

- Customized service profiles allow you to create your own profiles for calls through your H.323 Gatekeeper, Gateway, or SIP Proxy
- Multisession calls
- 3-way conferencing using different VoIP protocols
- Manual codec selection
- Support for extended H.323 address syntax
- Support for advanced SIP URI syntax
- Support for CallTo URL
- Lost and out-of-order packet indication
- Log window for operation monitoring
- Media Engine statistics
- Per-Call RTP statistics
- QoS/TOS/DiffServ support
- Advanced lost packet recovery, offering better sound quality over a poor connection.
- Optimized processor usage for HT/Multicore processors

Remote support console

- An SDK for developing third-party VoIP applications

Attention! Advanced lost packet recovery is not fully H.323-compatible. Use it only while communicating with other **SJphone™**-compatible software.

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1.2. SJphone™ Features Coming Soon

- Video
- Session recording
- Speex codec

To learn more about these features and to obtain the latest version of **SJphone™**, go to the SJ Labs web page at: http://www.sjlabs.com

1.3. System Requirements

A PC running Windows ME/2000sp4/XPSP2:

Hardware requirement:

- At least Intel Celeron 300 CPU with 32MB for Windows ME, 128MB for Windows 2000, 256 for Windows XP.
- A full-duplex sound card with a microphone and speakers, or any USB phones. Installed latest **DirectX** (9.0 at least) is strongly recommended; you can check it by `dxdiag.exe`. Please use latest drivers for soundcards and USB phones.
- Any kind of broadband TCP/IP connection like wired or wireless (802.11x, 802.16x) Ethernet or wide area connections (WAN) like DSL, E1/T1, etc are recommended. You will get a better sound quality with a wideband connection but a v.90 analog modem is enough if you are using voice compression with an appropriate codec.
- If you need to place a call to/from PSTN, you must have an ITSP account or access to an H.323 or SIP Gateway like CISCO 26xx/36xx/3700/53xx, etc. Refer to your gateway manual to check whether your gateway is configured correctly.

1.4. Installing SJphone™

Attention Win2000/XP users! You must have administrative privileges to install **SJphone™**. If you are not sure whether you have such privileges, you almost certainly do not have them. Contact your system administrator for assistance.

Attention Windows 2000! You need to install Microsoft Installer 2.0 from www.microsoft.com before installing **SJphone™**.

To install **SJphone™**

- Download and run the installation file.
- Follow the on-screen instructions. .

To uninstall **SJphone™**, go to **Control Panel**->**Add/Remove Programs**->**SJphone**->**Install/Uninstall**.

To start **SJphone™**, go to **Start**->**SJ Labs** program group->**SJphone**->**SJphone**.

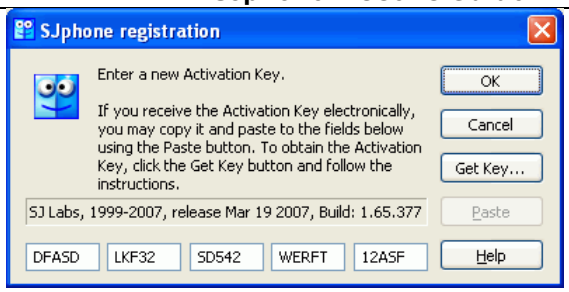
1.5. Registering SJphone™

You may register your **SJphone™**. The registration is free and removes the advertising pop-up window. You will also be able to receive information about new **SJphone™** releases.

To register your SJphone™ on-line,

- Click the **Options** button on the **SJphone™ Main** panel. The **Options** panel will appear.
- Select the **Support** tab, then click the **Register** button.

An **SJphone registration** message will appear.


SJphone registration
Enter a new Activation Key.
If you receive the Activation Key electronically, you may copy it and paste to the fields below using the Paste button. To obtain the Activation Key, click the Get Key button and follow the instructions.
SJ Labs, 1999-2007, release Mar 19 2007, Build: 1.65.377
DFASD
LKF32
SD542
WERFT
12ASF
OK
Cancel
Get Key...
Paste
Help

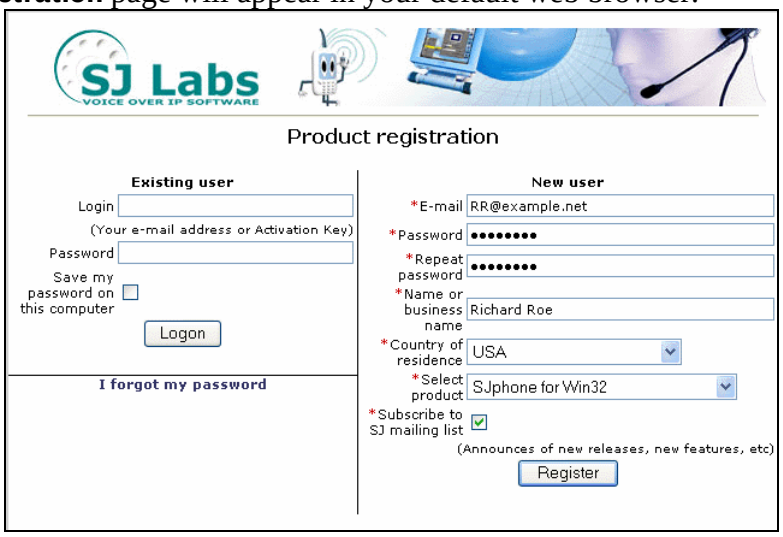
- Enter your activation key and click the **OK** button. The activation key is case-insensitive.

If you receive the key electronically, you may copy and paste it to the **SJphone registration** message using the **Paste** button.

If you do not have the activation key yet,

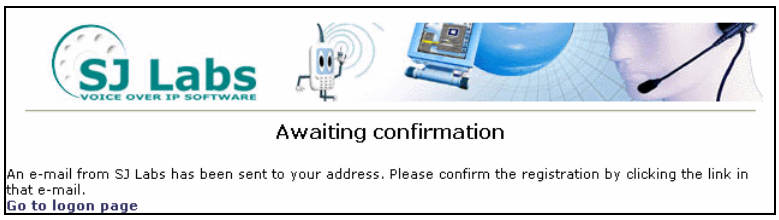
- Click the **Get Key** button.

The **Product Registration** page will appear in your default web browser.



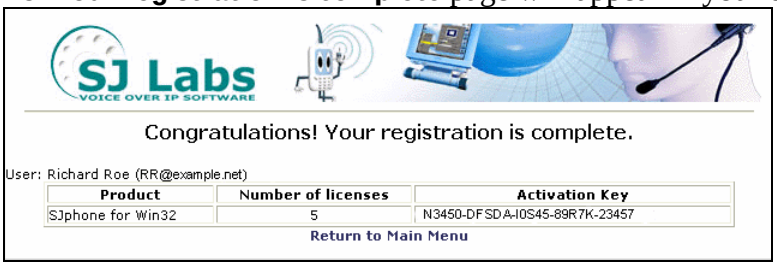
- Enter the necessary information and click the **Register** button.

A confirmation e-mail will be sent to your E-mail and the **Awaiting Confirmation** page will appear in your default web browser.


SJ Labs
VOICE OVER IP SOFTWARE
Awaiting confirmation
An e-mail from SJ Labs has been sent to your address. Please confirm the registration by clicking the link in that e-mail.
Go to logon page

- When the mail arrives, click a link in the mail.

The **Congratulations! Your registration is complete** page will appear in your default web browser.



- Copy and paste the activation key to the **SJphone registration** message and click the **OK** button.

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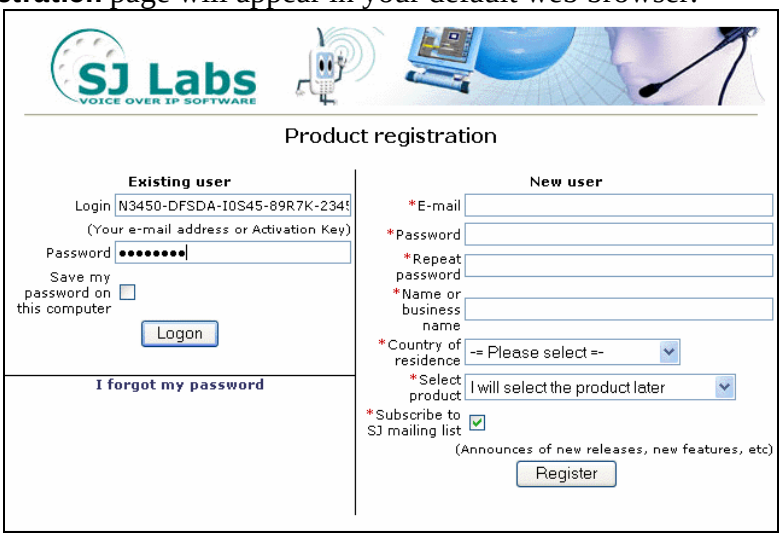
If you want to change your personal information

- Click the **Options** button on the **SJphone™ Main** panel. The **Options** panel will appear.
- Select the **Support** tab, then click the **Register** button.

An **SJphone registration** message will appear.

- Click the **Get Key** button.

The **Product Registration** page will appear in your default web browser.

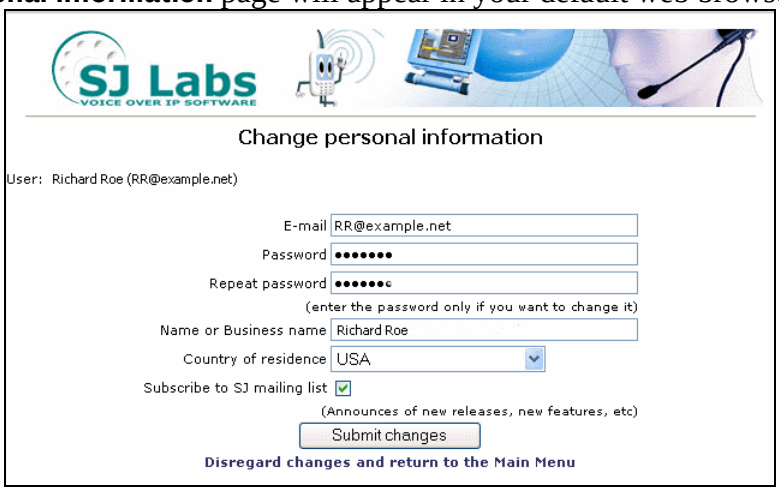


- Enter your Login and Password and click the **Logon** button. The Login is either an activation key of an SJ Labs registered product or your e-mail address used for the registration. If you come to this page from an already registered product, its activation key will be entered into Login automatically.

The **Main Menu** page will appear in your default web browser.

- Click **Change personal information**.

The **Change personal information** page will appear in your default web browser.



- Change required information and click the **Submit changes** button.

If you forgot your password,

- Click the **Options** button on the **SJphone™ Main** panel. The **Options** panel will appear.
- Select the **Support** tab, then click the **Register** button.

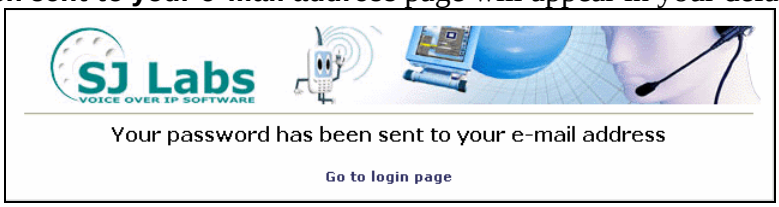
An **SJphone™ registration** message will appear.

- Click the **Get Key** button.

The **Product Registration** page will appear in your default web browser.

- Click **I forgot my password**.

An e-mail with the password will be sent to the e-mail address used for registration and the **Your password has been sent to your e-mail address** page will appear in your default web browser.



1.6. Contact Information and Technical Support

Email: `SJphone@sjlabs.com`
Web: **`http://www.sjphone.org`**

or

- Click the **Option** button on the **Main** panel, select the **Support** tab, and click the **Web** button.

Bug Reporting

This form is used to provide **SJphone™** developers with the information necessary to fix problems that may occur. You may automatically record and submit the information about the problems.

To record and submit a bug report,

- Right-click the **SJphone™** system tray icon or skin, select **Bug Reporting**, and then **Start Recording.**
- Reproduce the problem
- Right-click the **SJphone™** system tray icon or skin and select **Submit...**.

The **Submit Bug Report** window will appear

- Describe the problem in the window.
- Submit the bug report either manually or automatically using your default mail program.

You may cancel bug report by selecting **Cancel Recording**.

Remote Console

(available in the **Advanced Mode** interface only and may be unavailable for some service profiles)

Attention! You need to contact the support team to obtain the information on how to obtain remote on-line support.

Remote on-line support allows the support team to solve your problem remotely. They may review configuration options and immediately correct them if necessary. You need to contact the support team to obtain the information how to receive the remote on-line support.

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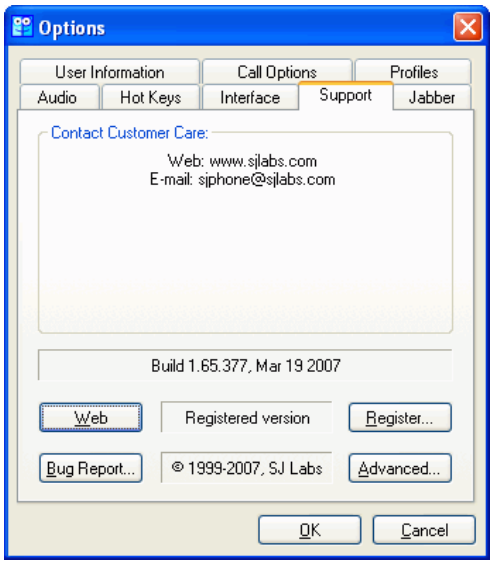
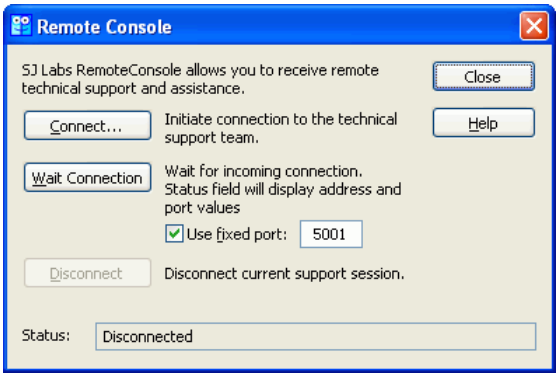


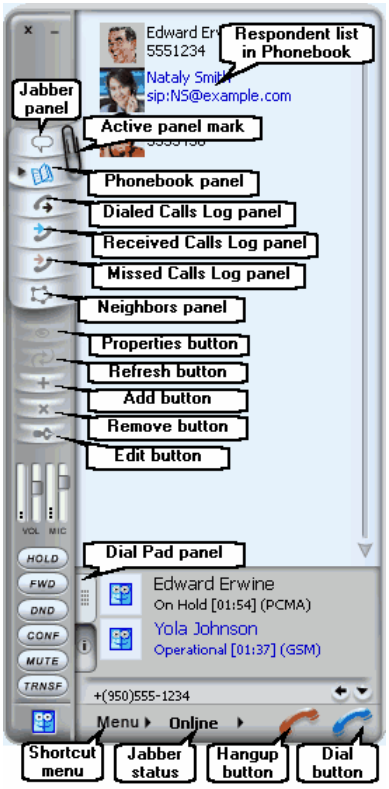
Web	Click this button to visit **SJ Labs** home page
Bug Report	Click this button to generate an e-mail bug report.
Register	Click this button to **register** your copy of **SJphone™**
Advanced	Click this button to activate **Remote Console**



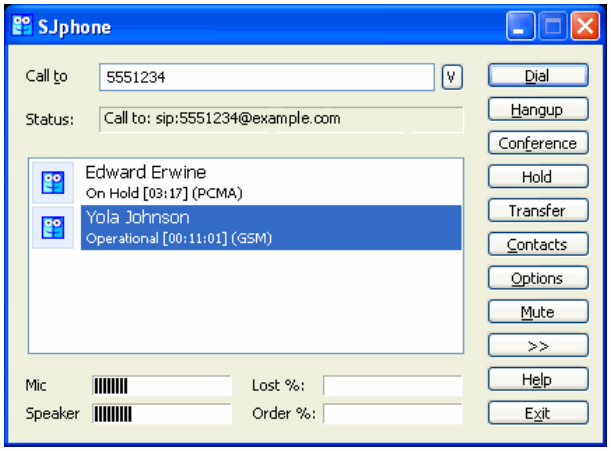
II. SJphone™ Panels and User Features

2.1. Controls on the Main Panel

Attention! Some **SJphone™** skins do not have the **Options** button. In this case, you can go to options by right-clicking the **SJphone™** system tray icon or skin and selecting **Options**.



The standard **Main** panel has two screens. Some skins may have more.



Call to:	Type here the telephone number, IP address, host name, or nickname you want to call
Status:	Displays the call status
Mic	Indicates input sound level for the current session
Speaker	Indicates output sound level for the current session
Lost%:	Indicates how many packets are lost for the current session
Order%:	Indicates how many packets are out of order for the current session
Buttons	
Dial	Click this button to dial the address typed in the **Call to**: field *Shortcut key:* **Alt** + **d** or **Enter**
Hangup	Click this button to end the call *Shortcut key:* **Alt** + **h** or **Enter** when in a call
Conference	Click this button to join selected respondents to a conference talk
Hold	Click this button to put the call on hold
Transfer	Click this button to transfer the call to the address specified in the **Call to**: field
Contacts	Click this button to activate the **SJphone™** Contacts
Voice Mail	Click this button to receive your voice mail.
Options	Click this button to activate the **SJphone™ Options** panel. Using this panel, you can configure **SJphone™**. *Shortcut key:* **Alt** + **o**
Mute Speak	Click the **Mute** button to mute your voice. The **Speak** button will appear. Click the **Speak** button to resume the conversation. The **Mute** button will appear again. *Shortcut key:* **Alt** + **m** and **Alt** + **s**
>>	Click this button to go to the next screen of **SJphone™ Main** panel
<<	
Exit	Click this button to close **SJphone™** into the system tray *Shortcut key:* **Alt** + **x**
Help	Click this button to start **SJphone™**'s **Help** *Shortcut key:* **Alt** + **e**

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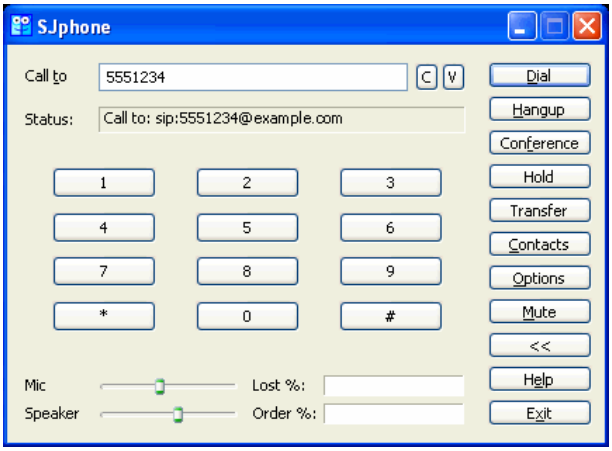
Sliders

Mic — Use this slider to adjust microphone volume

Speaker — Use this slider to adjust output sound volume



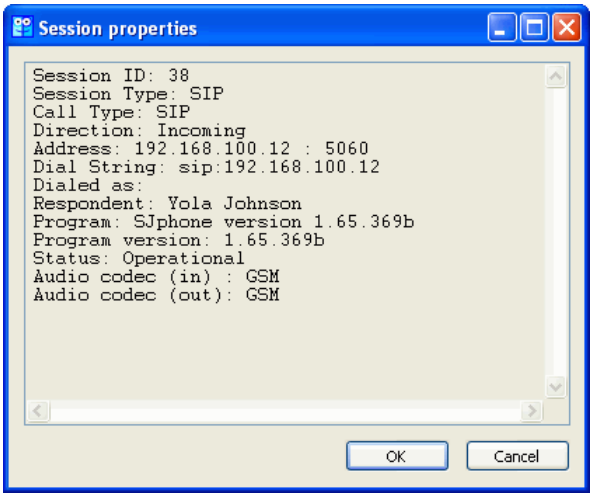
If you want to use **Speed Dial**, or send DTMF signals, you need to activate the dial pad screen by pressing the **>>** button. Use the ***** button to enter the dot. Use the **C** button to enter backspace.

You can select the **Main** panel appearance from available skins. Go to the Skins section to learn more about this feature.

You may look at the respondent properties.

(available in the **Advanced Mode** interface only and may be unavailable for some service profiles)

Right-click the respondent and select **Properties** in the shortcut menu. The **Session Properties** message will appear.



Lost and Out-of-Order Packets:

For quality communication, the number of lost packets should not exceed 10-15%. If the number of lost packets is too high, go to the Connection issues section for troubleshooting.

A large number of out-of order packets means that your network is overloaded or unstable. Congestion in packet switched networks can cause packets to take different routes to reach the same destination. Packets may arrive in the wrong order resulting in garbled speech. To prevent this, **SJphone™** stores received packets in a special buffer and reorders them. That is why even a large number of out-of-order packets does not always deteriorate the sound quality if the packets arrive within a certain time. If a packet comes too late, it gets lost.

Sound Adjustments:

Generally, you can adjust the **SJphone™** sound characteristics using standard Windows audio controls or controls on the **Main** panel.

The **Mic** and **Speaker** controls on the **Main** panel are connected with the standard Windows audio controls. Any changes in their settings automatically change the corresponding settings on the Windows volume control panel.

Some additional audio options may be set on the **Audio** tab.

2.2. Contacts

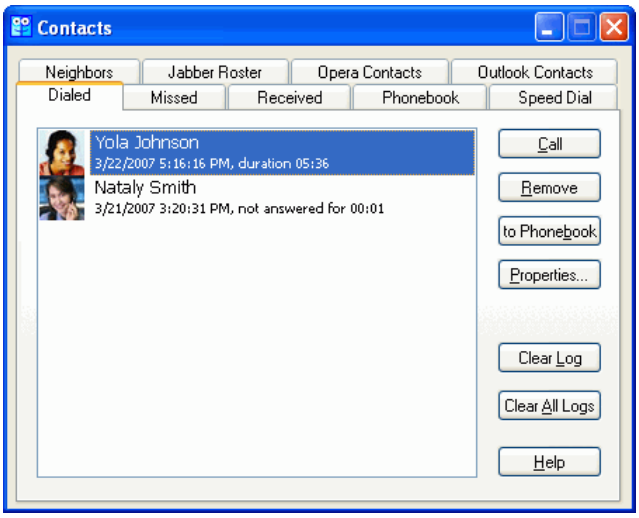
Contacts allow you to store information about respondents you may want to call regularly. They also have **Neighborhood** browser and logs for missed, received, and dialed calls.

To reach Contacts**,**

- Click the **Contacts** button
  or
  Right-click the **SJphone™** system tray icon or skin and select **Contacts** on the shortcut menu.

Call logs (Dialed, **Missed**, and **Received)**

You may see the list of missed, received, and dialed calls. You may repeat calls and see call records.



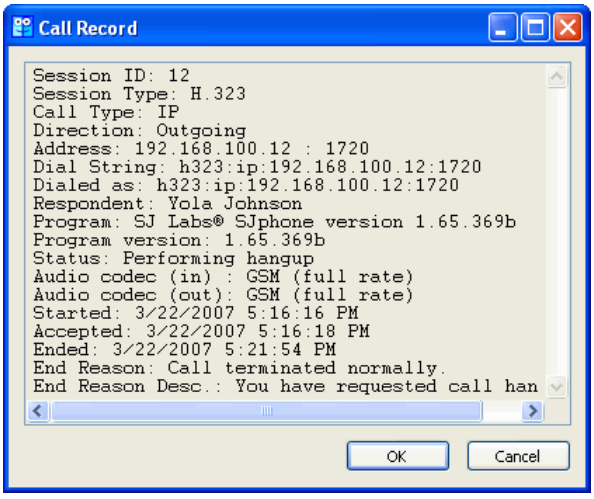
Call	Click this button to call the selected respondent
Clear Log	Click this button to remove all records in this log.
Properties	Click this button to view the information about a selected respondent
Remove	Click this button to remove the selected record from this log
to Phonebook	Click this button to add a selected respondent to **Phonebook**
Clear All Logs	Click this button to clear all logs.
Help	Click this button to activate **SJphone™ Help**

You may view the information about a call

(available in the **Advanced Mode** interface only and may be unavailable for some service profiles)

To do this, double-click the call, or click the **Properties** button.

The **Call Record** message will appear.



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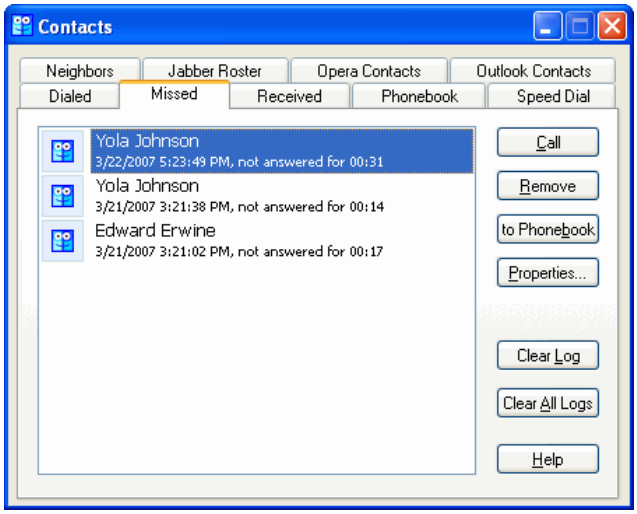
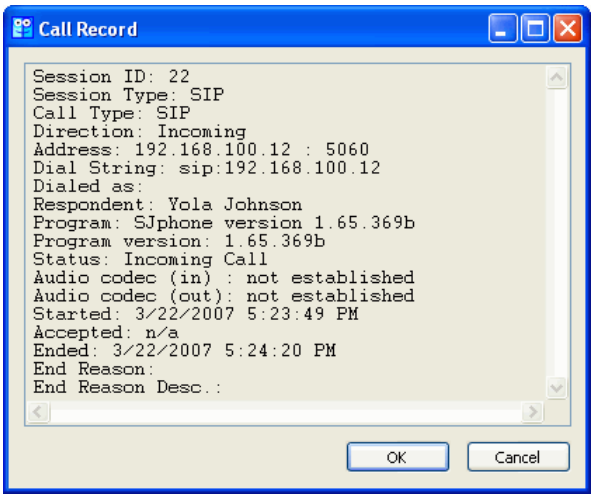
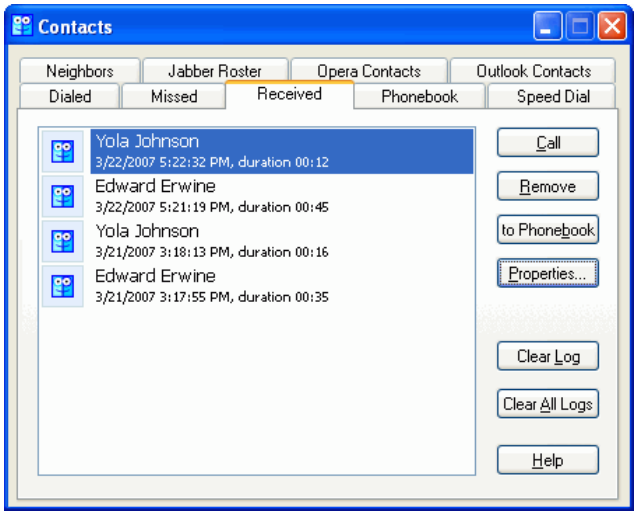
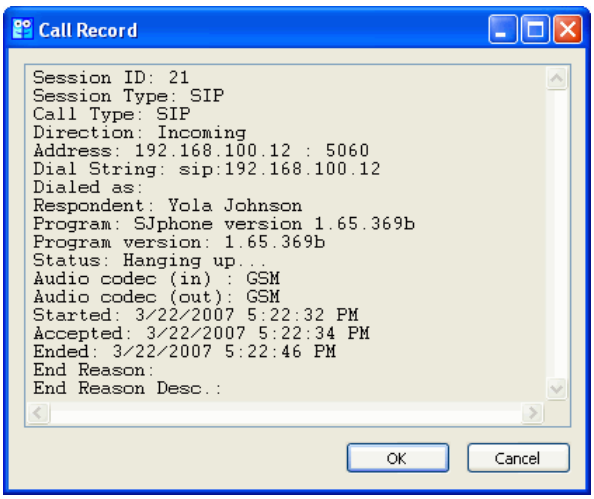


Call Record message





Call Record message



To repeat a call,

- Select the call record and click the **Call** button,
  or
  Double-click the call record.

To view the information about a call

(available in the **Advanced Mode** interface only and may be unavailable for some service profiles)

- Click the **Properties** button

The **Call Record** tab will appear.

To remove a call record,

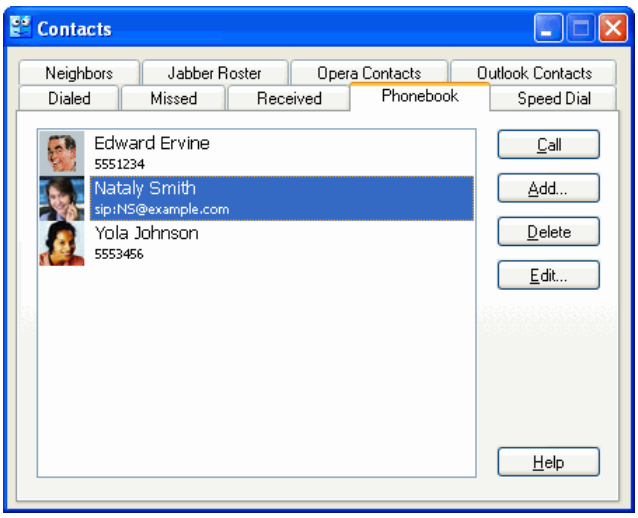
- Select the record and click the **Remove** button

To clear the logs,

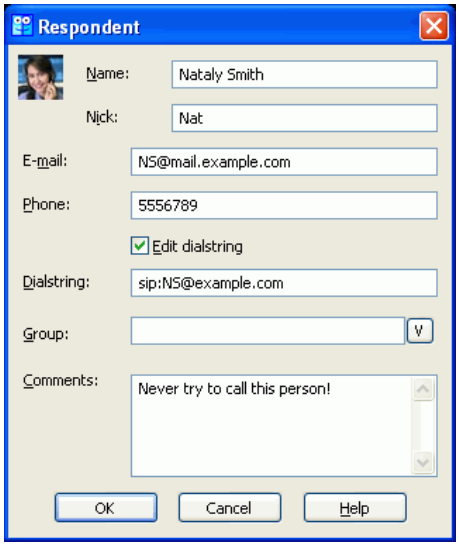
- Click the **Clear All Logs** button.

Phonebook

You can use the **Phonebook** panel in **Contacts** to store information about respondents whom you want to call later. You may store either a phone number or an advanced dial string which may be any valid dial string. See the Place a Call section for details.



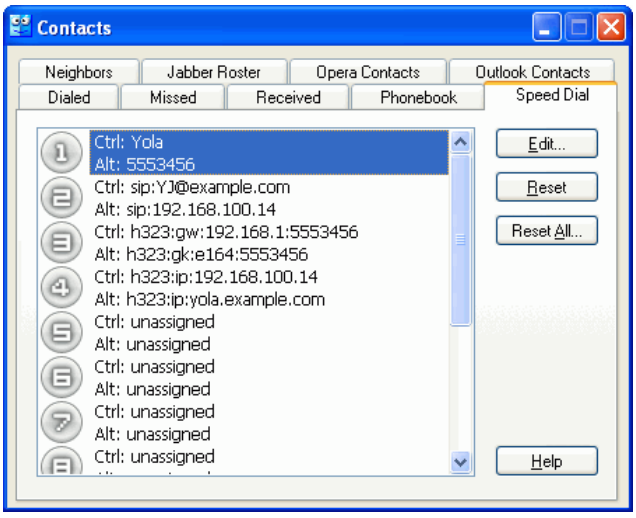
Call	Click this button to call the selected respondent
Add	Click this button to add a new record to **Phonebook**
Delete	Click this button to remove the selected record from **Phonebook**
Edit	Click this button to edit the selected record
Help	Click this button to activate **SJphone™ Help**



Full name:	Field for the respondent's full name
Nick name:	Field for the respondent's nickname
E-mail:	Field for the respondent's e-mail
Phone:	Field for the respondent's phone number
Edit Dialstring	Select this box if you want to use an advanced dial string. Go to the Place a Call section for details
Dialstring:	Field for the advanced dialstring for the respondent.
Group:	Field for respondent group
Comments:	Field for comments

Speed Dial

Press Alt/Ctrl and click the necessary button on the main panel, or press the digital button on your keyboard. **SJphone™** will place the information in the **Call To:** field.



Edit	Click this button to edit the selected record. The **Speed Dial Key** window will appear.

Speed Dial Key "1"
Provide name or address to be dialed:
Ctrl + <key>:
Yola
Alt + <key>:
5553456
Note: if your handset supports speed dialing then the address will be picked up from Ctrl + <key> field.
OK
Cancel
Help

Reset	Click this button to delete the information in the selected record
Reset All	Click this button to delete the information in all records
Help	Click this button to activate **SJphone™ Help**

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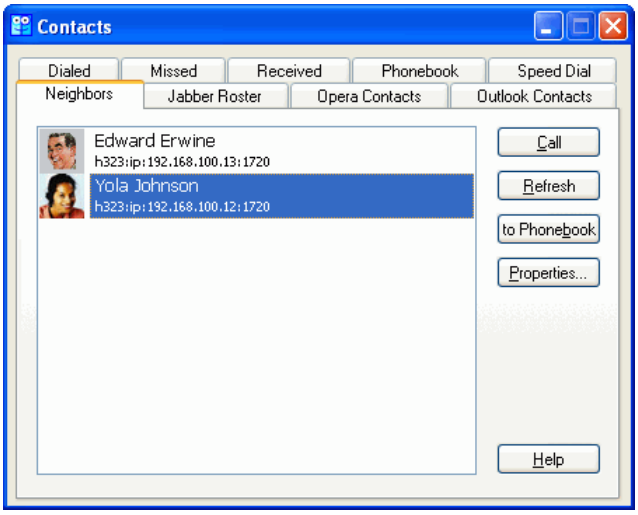
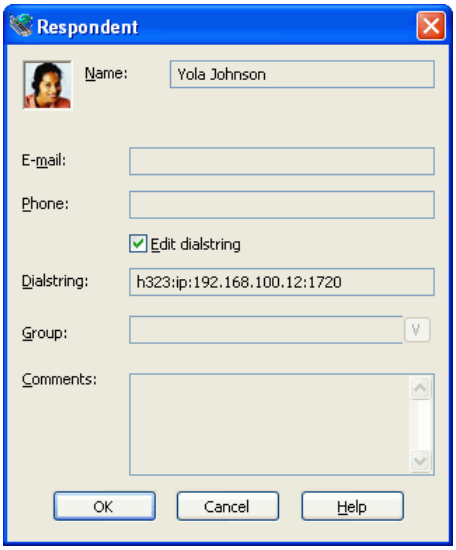
Neighbors

You may see other persons who are using **SJphone™** on your local network. You may call them or add to **Contacts**.

Attention! This feature works only on local area networks! If you use a personal firewall on your computer, make sure that it does not block UDP port 5003.



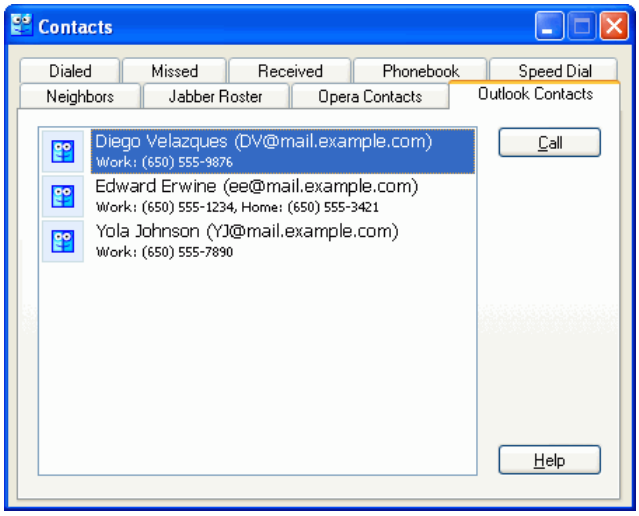
Call	Click this button to call a selected neighbor
Refresh	Click this button to refresh the list of neighbors
to Phonebook	Click this button to add a respondent from the **Neighbors** list to **Phonebook**
Properties	Click this button to view the information about a selected neighbor
Help	Click this button to activate **SJphone™ Help**



Outlook Contacts

Shows the contacts in the Microsoft Outlook or Outlook Express with phone numbers. It appears only if Microsoft Outlook or Outlook Express is installed and opens the contacts from that Outlook which is set as the e-mail program in the Windows Internet options.

To call the phone number of a selected contact, you need to select a profile for telephone calls.



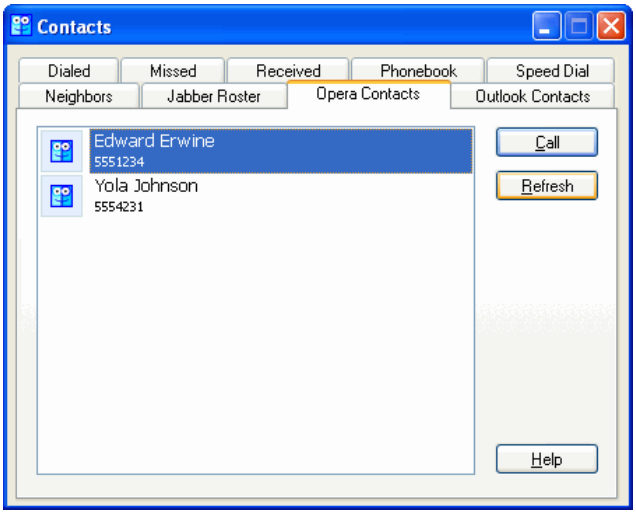
Call — Click this button to call a selected neighbor

Help — Click this button to activate **SJphone™ Help**

Opera Contacts

Shows the contacts in the Opera browser with phone numbers or e-mail addresses. It appears only if the Opera browser is installed.

To call the phone number of a selected contact, you need to select a profile for telephone calls.



Call — Click this button to call a selected contacts

Refresh — Click this button to refresh the list of Opera contacts

Help — Click this button to activate **SJphone™ Help**

Jabber Roster

(available in the **Advanced Mode** interface only and may be unavailable for some service profiles)

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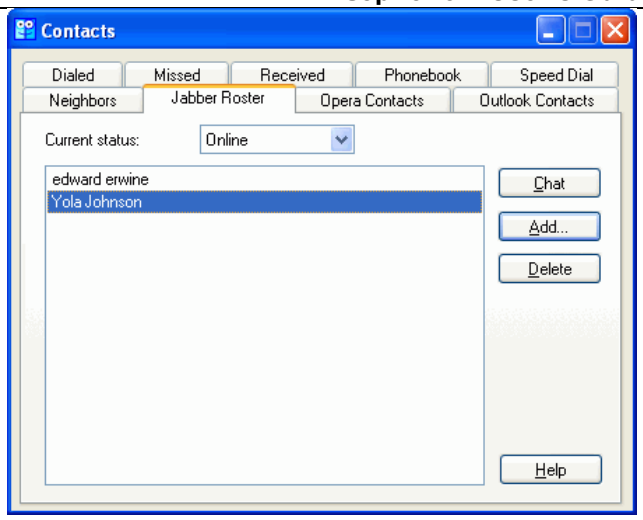


Chat	Click this button to start a chat with a selected user
Add	Click this button to add a respondent to the list of contacts
Delete	Click this button to delete a respondent to the list of contacts
Help	Click this button to activate **SJphone™ Help**

To connect SJphone™ to a Jabber server,

- select **Connect** in the **Current status** list.

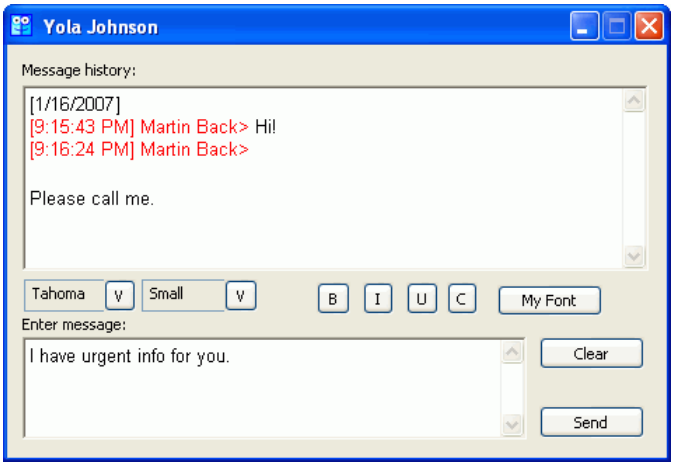
You may configure **SJphone™** to automatically connect to the Jabber server at each startup. See the Jabber Options section for details.

To start a chat with a user,

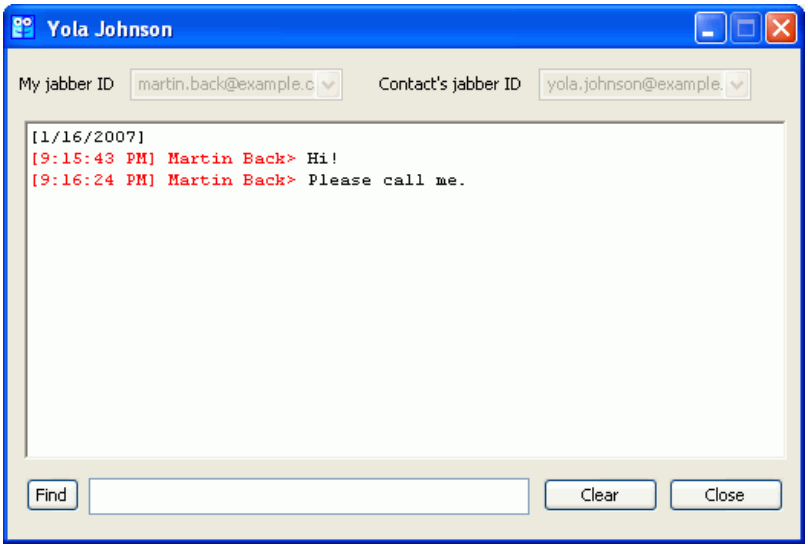
- select the user and click the **Chat** button.

The **Chat Session:** dialog box will appear.

You will see the chat history with that user and may enter your message. When you finish the message, click the **Send** button or press Enter on the keyboard. Press Shift+Enter to start a new line in the message



When another user sends you a message, **SJphone™** will show you a shortcut message. Double-click the message and answer the user.



To view chat history for a user,

- double-click the user and select **History**.

The **Message History** dialog box will appear.

You may clear the history by clicking the Clear button.

To change your status on the Jabber server,

- select the status in the **Current status** list.

To add a user to a roster,

- click the **Add...** button and enter the user's Jabber ID on the **Add new Jabber contact** dialog box.

The user will be notified that you want to add it to your roster. If the user allows you to add it to the roster, you will receive the notification message, and its ID will appear on the Jabber roster.

To delete a user from your Jabber roster,

- select the user and click the **Delete** button.

The user will be notified that you deleted it from your roster.

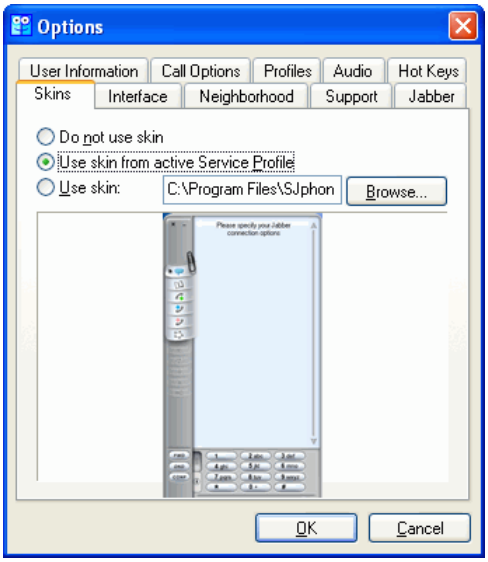
To disconnect SJphone™ from the Jabber server,

- select **Disconnect** in the **Current status** list.

2.3. Skins

You can choose the **SJphone™ Main** panel appearance by using skins.

Attention! Some skins may have no **Options** button. To access the **Options** panel, right-click the **SJphone™** system tray icon or skin and select **Options** on the shortcut menu.



Do not use skin	Select this option if you do not want to use skins
Use skin from active Service Profile	Select this option if you want to use a skin specified in the active **service profiles**
Use skin	Select this option if you want to use a skin that you have selected
Browse	Click this button to browse for a skin

To change the skin,

- Click the **Options** button on the **Main** panel.
- Go to the **Skins** tab, select **Use skin**, and click the **Browse** buttons to find the skin you prefer from the list.

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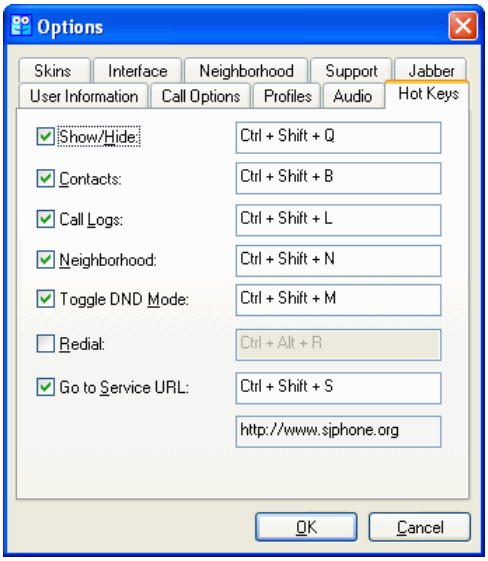
2.4. Hot Keys

You may assign several hot keys to perform common operations on the **Hot Keys** tab.



Attention! When any of the **Option** tabs is active, the hot keys are disabled.

2.5. Vertical Service Codes

(May be unavailable for some service profiles)

SJphone™ supports a number of special commands that can be issued by dialing special prefixes or numbers. These commands are standardized as Vertical Service Codes and also known as Star Codes in many PBX systems (for more information, see the NANPA site).

You may enable and disable the codes and control the way **SJphone™** treats the codes on the **Advanced** tab in each SIP profile.

- **To enable the Vertical Service Codes**, select the Enable service codes option.
- **To make SJphone™ remove** the Vertical Service Codes before sending an entered number to the server, select the Remove service codes option.

SJphone™ supports the following Vertical Service Codes:

The Do Not Disturb mode:

To turn the **Do Not Disturb** mode on: ***78**
To turn the **Do Not Disturb** mode off: ***79**

If the Do Not Disturb mode is turned on, **SJphone™** will reject all incoming calls. **SJphone™** does not send these codes to the server

Missed Calls: *51

If this code is dialed, **SJphone™** will show the **Missed Calls** panels from **Contacts**. **SJphone™** does not send this code to the server.

Anonymous call rejection

To turn **Anonymous call rejection** on: ***77**
To turn **Anonymous call rejection** off: ***87**

If Anonymous call rejection is turned on, **SJphone™** will automatically reject all anonymous calls as if they were rejected manually. **SJphone™** does not send these codes to the server

Ignoring all incoming calls during one call: *70<number to dial>

If this code is added as a prefix to the number to dial, **SJphone™** will reject all incoming calls during that call. As soon as that call ends, **SJphone™** will continue to accept calls in the way set on the **Call Options** tab. This code can be used if you make an important call and do not want to be interrupted.

Restoring Caller ID delivery: *82<number to dial>

If this code is added as a prefix to the number to dial, **SJphone™** will show your Caller ID even if Block Caller ID delivery is selected on the **Call Options** tab.

Changing Caller ID delivery: *67<number to dial>

If this code is added as a prefix to the number to dial, **SJphone™** will change the Block Caller ID delivery option specified on the Call Options tab to the opposite.

That is, if the Block Caller ID delivery option is selected, this code will make **SJphone™** show your Caller ID for that call only, or if the Block Caller ID delivery option is cleared, this code will make **SJphone™** hide your Caller ID for that call only.

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III. Configuring SJphone™

This chapter explains how to configure **SJphone™** basic settings.

Attention! When any of the **Options** tabs is active, **SJphone™** automatically rejects all incoming calls.

Attention! Some tabs are not available on the default Simple Mode interface. To reach them, you need to switch to the Advanced Mode.

Interface mode switch:

- Right-click the **SJphone™** system tray icon or skin,
- Select either **To Advanced Mode** or **To Simple Mode**

Attention! The Advanced Mode may be unavailable for some service profiles.

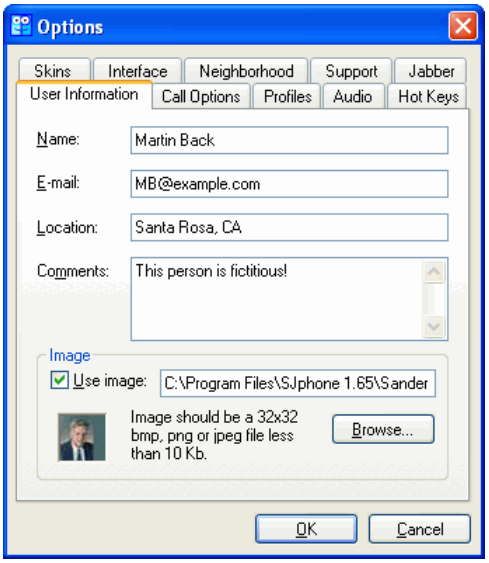
You can reach **SJphone™** configuration controls by clicking the **Options** button on the **Main** panel, or by right-clicking the **SJphone™** system tray icon or skin and selecting **Options** in the shortcut menu.

Attention! Some **SJphone™** skins do not have the **Options** button. In this case, right-click the **SJphone™** system tray icon or skin.

3.1. Entering Your Personal Information

You need to personalize your **SJphone™** copy when you start your **SJphone™** for the first time. You also may change this information any time you like.

Note: This information will be available to your respondents while you are on a call.



Name	Field for your name
E-mail:	Field for your e-mail
Location:	Field for your location
Comments:	Field for comments
Image	You may add a picture to your personal information. All other **SJphone™** users will see it on their main panel.
Use image	Select this box if you want other **SJphone™** users to view your picture
Browse	Click this button to browse for your picture

To personalize your SJphone™ copy,

- Click the **Options** button on the **SJphone™ Main** panel.

The **User Information** tab will appear.

- Enter your personal information and click the **OK** button.

Note: Some ITSP require that the Name on the **User Information** tab should coincide with Account or Caller ID entered during **profile personalization**.

You may add a picture to your personal information. The picture should be 32x32 pixels. All other **SJphone™** users will see it on their main panel if the H.323 VoIP protocol is used.

SJphone™ will use unknown as the username if you do not specify it in the Name field, or this field is cleared and left blank.

Note: This information will be available to your respondents while you are on a call.

3.2. Call Options

You may set call options on the **Call Options** tab.

Incoming Calls:

You can configure your **SJphone™** to automatically accept or reject all incoming calls.

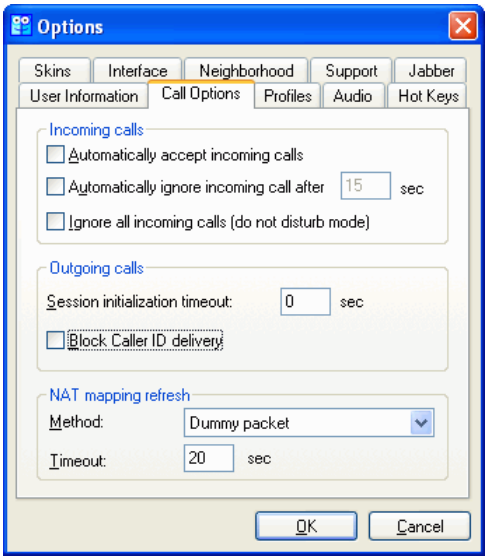
Only one of the Automatically accept incoming calls, Automatically ignore incoming call after… secs, or Ignore all incoming calls checkboxes can be selected at a time.

Alternatively, you can turn **Auto accept calls** or **Do not disturb mode** on and off on the **SJphone™** shortcut menu. Right-click the system tray icon or skin to access this menu.

Outgoing Calls:

You may **Block Caller ID delivery** to conceal your name, username, and other sensitive information during a call as much as possible. If this option is selected, your call will come as `Anonymous` from the `sip:anonymous@anonymous.invalid` address. Please note that the policy of accepting/delivering anonymous calls varies for service providers.

You may turn this option specifically for a call on the shortcut menu. Right-click the system tray icon or skin and select **Block Caller ID delivery** on the shortcut menu.



Incoming calls:

Automatically accept incoming calls:	Select this box if you want **SJphone™** to accept all incoming calls automatically
Automatically ignore incoming call after… secs:	Select this box if you want **SJphone™** to automatically ignore an incoming call if you do not accept it within the specified time
Ignore all incoming calls:	Select this box if you want **SJphone™** to ignore all incoming calls

Outgoing calls:

Session initialization timeout	Time interwal for which **SJphone™** waits for registration with a SIP proxy. If it fails to register during this timeout, the session will not start. `0` disables this feature.
Block Caller ID delivery	Select this box to conceal your name, username, and other sensitive information during a call as much as possible.

NAT mapping refresh

Method	If your service provider supports Dummy packets, select it to reduce server load. If not, select Option request. Always consult your service provider/network administrator if in doubt.
Timeout	Time interval in seconds in which **SJphone™** sends the OPTIONS requests. `0` makes **SJphone™** stop sending the requests. Do not change its default value (`20`) unless it is absolutely necessary.

Anonymous calls

You may set **SJphone™** to automatically reject all calls from anonymous respondents. Right-click the system tray icon or skin and select **Reject anonymous calls** on the shortcut menu.

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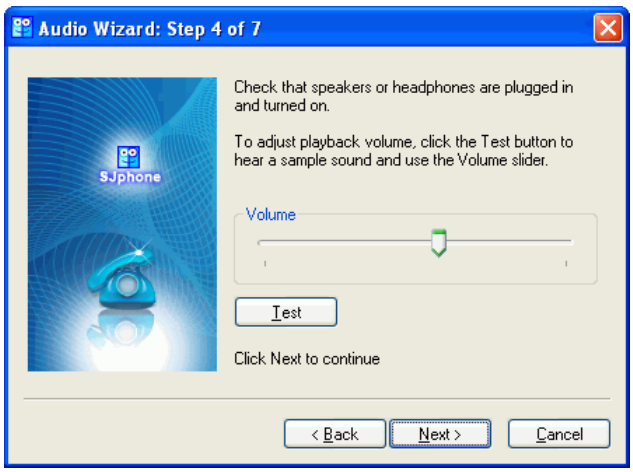
3.3. Audio Options

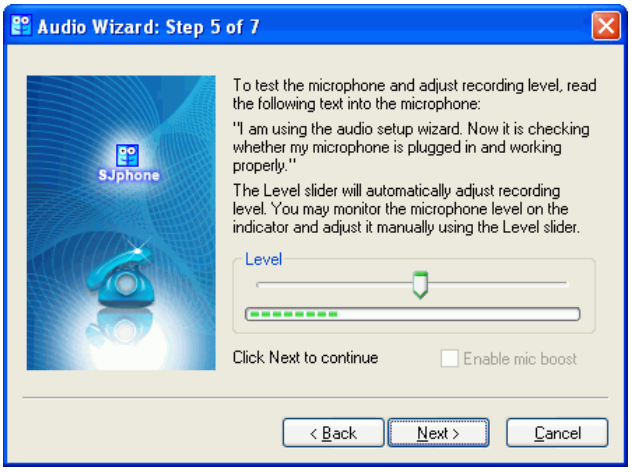
Audio Wizard

The best way to adjust **SJphone™**'s audio settings is to use a build-in Audio Wizard. It starts automatically on the first **SJphone™** startup and will guide you through the audio setup procedure. If necessary, you may start the Audio Wizard later manually. Right-click the system tray icon or skin, select Audio Wizard on the shortcut menu, and follow the on-screen instructions.

Audio Wizard: Step 1 of 7
This wizard helps you to tune the softphone audio settings.
To start tuning:
1. Close all other programs that play or record sound.
2. Make sure that speakers and microphone are plugged in and turned on.
Click Next to continue
< Back
Next >
Cancel

Audio Wizard: Step 2 of 7
Use DirectX for the best performance.
DirectX improves audio performance by shortening the time interval between the moment audio is sent and the moment you receive it.
DirectX decreases sound latency but may cause problems with some sound cards.
You have DirectX version 9.0a or newer installed.
Use DirectX (DirectSound)
Click Next to continue
< Back
Next >
Cancel


Audio Wizard: Step 4 of 7
Check that speakers or headphones are plugged in and turned on.
To adjust playback volume, click the Test button to hear a sample sound and use the Volume slider.
Volume
Test
Click Next to continue
< Back
Next >
Cancel


Audio Wizard: Step 5 of 7
To test the microphone and adjust recording level, read the following text into the microphone:
"I am using the audio setup wizard. Now it is checking whether my microphone is plugged in and working properly."
The Level slider will automatically adjust recording level. You may monitor the microphone level on the indicator and adjust it manually using the Level slider.
Level
Click Next to continue
Enable mic boost
< Back
Next >
Cancel

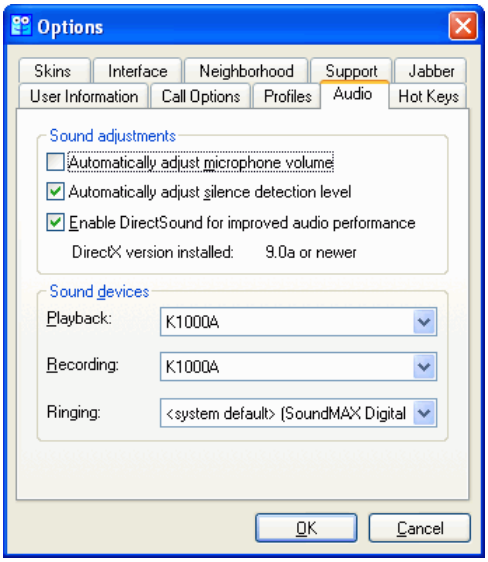
Audio Wizard: Step 6 of 7
Verify that the audio settings are adjusted correctly:
1. Click the Record button and say something into the microphone for 10 sec.
2. Click the Play button and listen to the recorded audio.
Record
Play
3. Click Next to continue if sound quality is good, otherwise click Back and return to the previous panels to repeat sound adjustment.
< Back
Next >
Cancel

Audio Wizard: Step 7 of 7
You have tuned your audio settings for the softphone.
Run this wizard again:
1. When you have troubles with audio.
2. When you have changed the phone playback or recording audio device.
3. When you have installed a new audio device, for example, a new USB handset.
Click Finish to continue
< Back
Finish
Cancel

Generally, you can adjust the **SJphone™** sound characteristics using standard Windows audio controls or controls on the **Main** panel.

The **Mic** and **Speaker** controls on the **Main** panel are connected with the standard Windows audio controls. Any changes in their settings automatically change the corresponding settings on the Windows volume control panel.

Some additional audio options may be set on the **Audio** tab.



Sound adjustments

Automatically adjust microphone volume	Select this box to let **SJphone™** adjust the microphone volume automatically
Automatically adjust silence detection level	Select this box to enable the automatic adjustment of silence detection level
Enable DirectSound for improved audio performance	Select this box if your sound card supports **DirectSound**
DirectX version installed:	Check **DirectX** version installed in your system

Sound devices

Playback:	Shows which audio device **SJphone™** uses to playback audio.
Recording:	Shows which audio device **SJphone™** uses to record audio.
Ringing:	Shows which audio device **SJphone™** uses to play a ringing sound.

Compression Settings:	Click this button to set **SJphone™** compression settings.
Advanced Settings:	Click this button to set **SJphone™** advanced audio settings.

Automatically adjust microphone volume

Some sound cards do not have the automatic microphone gain feature. You can select Automatically adjust microphone volume option to let **SJphone™** automatically adjust the microphone sound level.

Automatically adjust silence detection level

When the Automatically adjust silence detection level is selected, **SJphone™** does not compress silence reducing processor load. If there is a strong background sound when this feature is selected, **SJphone™** may not detect silence correctly, and fragments of your speech may be lost.

Enable Direct Sound for improved audio performance

Select Enable Direct Sound for improved audio performance if your sound card supports DirectSound. This will decrease sound latency time. Most sound cards from Turtle Beach, Yamaha, Creative Labs

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(excluding the Ensoniq-based AudioPCI types), Diamond, Crystal, or Microsoft USB speakers support this feature.

SJphone™ supports DirectX version 9.0 and higher.

Sound Devices

If your computer has several sound devices, you may select which one will be used to play incoming (Recording), ringing (Ringing), and record outgoing (Playback) sound. **SJphone™** uses a system default audio device to play audio during a call unless you manually select a specific audio device in this option. If the system does not find this device, **SJphone™** will use the system default audio device. When the system finds the specified audio device, **SJphone™** will use it again.

If you have installed a USB handset or other audio device:

A new audio device will appear in your system. Go to the **Audio** tab and select the device as the Playback and Recording device in Sound devices.

To use handset buttons and ringer with **SJphone™**, you need to download and install a special driver from the SJ Labs, Inc web site.

Compression and Advanced Audio Settings

(available in the **Advanced Mode** interface only and may be unavailable for some service profiles)

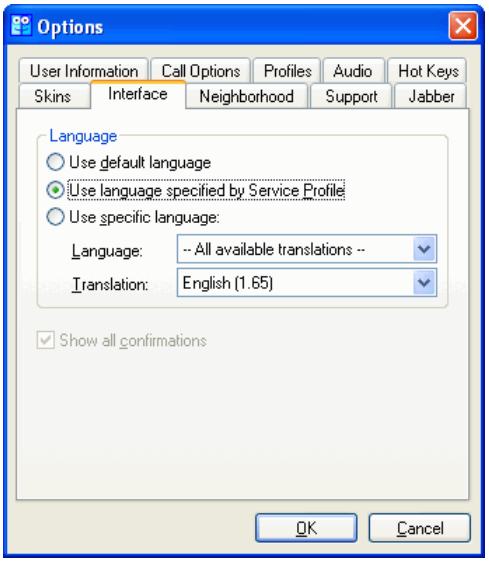
Advanced audio and compression settings such as Compression settings or SJ Labs Extensions are explained in detail in the **Compression and Advanced Audio Settings** section. Do not change them unless you completely understand what you are doing, or you may severely degrade the **SJphone™** performance.

3.4. Interface Options

You can change the language of **SJphone™** user interface on the Interface tab. You may select the following options:

1. The default language (English).
2. The language specified by the current service profile.
3. A language from a list of available language packs.

You may also make **SJphone™** show all confirmation messages.



Language	
Use default language	If selected, **SJphone™** will use its default interface language (English).
Use language specified by Service Profile	If selected, **SJphone™** will use the interface language specified in the current service profile.
Use specific language	If selected, **SJphone™** will use the interface language from a list of available language packs.
Show all confirmations	If this option is selected, **SJphone™** shows all confirmation messages.

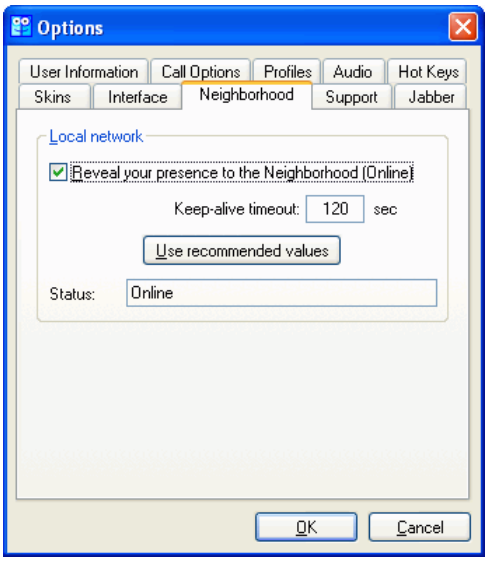
3.5. Neighborhood Options

(available in the **Advanced Mode** interface only and may be unavailable for some service profiles)

You may set your neighborhood parameters on the **Neighborhood** tab.

Attention! This feature works only on local area networks! If you use a personal firewall on your computer, make sure that it does not block UDP port 5003.

When running, **SJphone™** constantly sends broadcast messages to the local network. Other **SJphone™**s running on computers on this network receive these messages and add **SJphone™** to their **Neighborhood** list. The users of those computers may find your presence and call you. You may enable or disable this feature.



Local network

Reveal your presence to the Neighborhood (Online):	Check this box to reveal your presence to your network neighbors
Keep-alive timeout:	Field for the timeout in which **SJphone™** sends messages about its presence to other **SJphone™**s on your local network.
Status:	Shows your neighborhood status.

Buttons

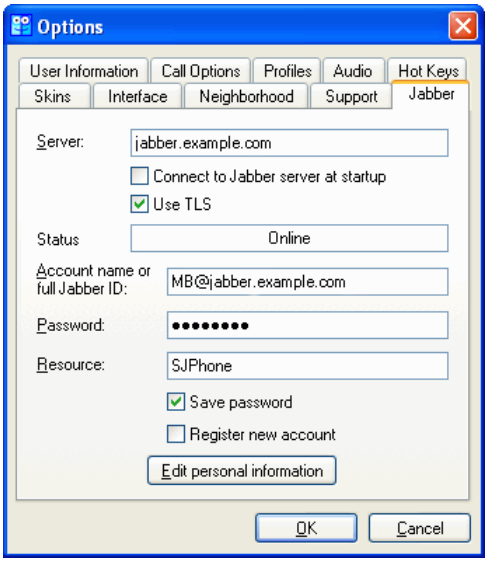
Use recommended values	Click this button to set the values to their recommended values

You may use recommended values for this procedure. Click the **Use recommended values** button.

3.6. Jabber Options

(available in the **Advanced Mode** interface only and may be unavailable for some service profiles)

SJphone™ supports the Jabber instant messaging technology. You may set Jabber parameters on the **Jabber** tab.



Server	Field to enter a Jabber server.
Connect to Jabber server at startup	Select this check box if you want **SJphone™** to automatically connect to the Jabber server at each startup.
Use TLS	Select this check box if you want to establish secure communication between **SJphone™** and the Jabber server.
Status	Shows the current status on the Jabber server.
Account name or full Jabber ID	Field to enter your account name or full Jabber ID.
Password	Field to specify your password.
Resourse	Indicates a Jabber client that is used to connect to the server. If you log to the Jabber servers simultaneously from several different Jabber clients under the same Jabber ID, you need to have a unique Resource for each Jabber client.

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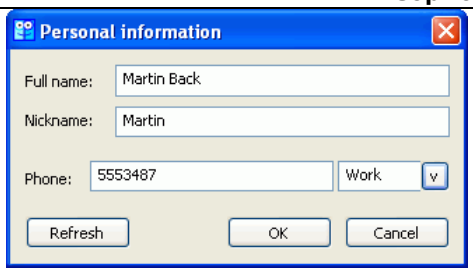


Save password — Select this check box if you want **SJphone™** to automatically apply the password.

Register new account — Select this check box if you want to register a new account at the Jabber server.

Buttons

Edit personal information — Click this button to edit your personal information.

3.7. Customized Incoming Call Sound

You may use an incoming call sound of your own choice if you rename the sound file into `ring.wav` and place it in the **SJphone™** folder, default is …`\Program Files\SJphone 1.65`.

You may also use another sound file named `beep.wav` as a second incoming call notification (it will be played if you have an active session). Place it in the same folder.

Those files should be in the PCM 8/16 kHz, 16 bit, mono format.

3.8. Settings for Firewall or Other Network Device

If you have a personal firewall installed on your computer, a number of ports should be opened for the `SJphone.exe` application both for inbound and outbound connections. Go to the Used Port section for details.

Refer to your personal firewall documentation for instructions how to make necessary adjustments.

At some firewalls, you may set `SJphone.exe` as a trusted application, that is, an application which is allowed to make any connection by default. Refer to your personal firewall documentation for instructions.

If your computer is on a home or private network, behind a corporate firewall, or connected to the Internet through a router with NAT, the same ports should be opened/forwarded on that network device. Refer to your network device documentation for details.

3.9. Windows XP Service Pack 2 Settings

Generally, **SJphone™** should work fine with minimum adjustments in Windows Security Center.

If you turn off the Windows firewall in Security Center, no settings are required to change for Windows or **SJphone™**.

If XP Service Pack 2 is installed on your computer with its default-enabled Windows firewall, the **Windows Security Alert** message will appear the first time you start **SJphone™**. This message will inform you that it has blocked **SJphone™** from accepting connections from the Internet or a network.

Windows Security Alert
To help protect your computer, Windows Firewall has blocked some features of this program.
Do you want to keep blocking this program?
Name: Softphone
Publisher: SJ Labs, Inc.
Keep Blocking
Unblock
Ask Me Later
Windows Firewall has blocked this program from accepting connections from the Internet or a network. If you recognize the program or trust the publisher, you can unblock it. When should I unblock a program?

To allow SJphone™ receive incoming calls,

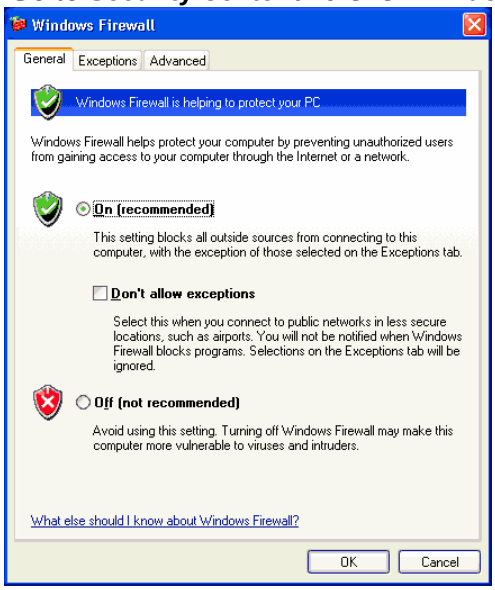
- Click the **Unblock** button on the message.

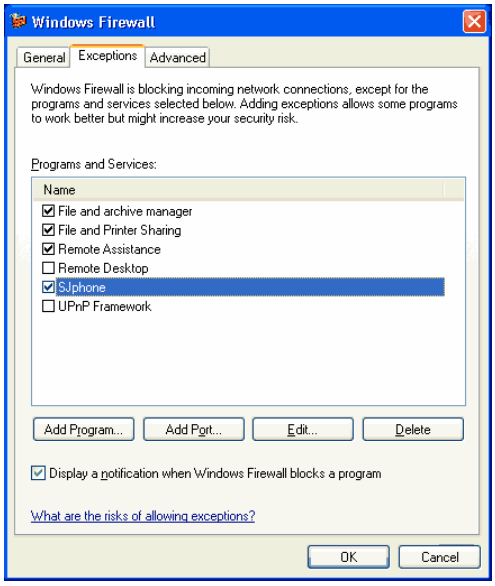
Now **SJphone™** will be able to receive incoming calls. No other additional settings are required for Windows or **SJphone™**.

If you have clicked the **Keep Blocking** button, **SJphone™** will not be able to receive incoming calls.

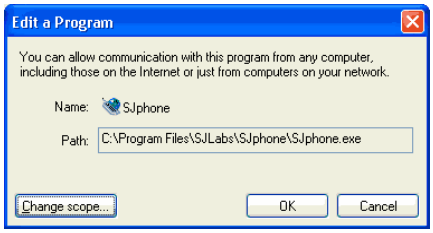
To allow SJphone™ receive incoming calls,

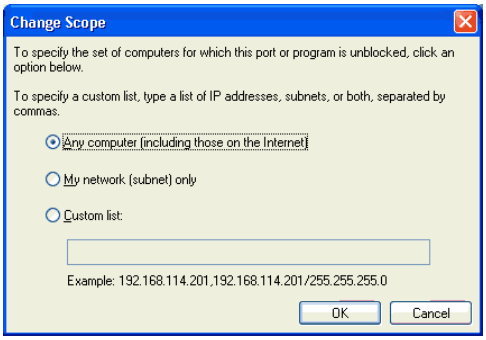
- Check that **SJphone™** is running.
- Go to **Security Center** and click **Windows Firewall**.


Windows Firewall
General
Exceptions
Advanced
Windows Firewall is helping to protect your PC
Windows Firewall helps protect your computer by preventing unauthorized users from gaining access to your computer through the Internet or a network.
On (recommended)
This setting blocks all outside sources from connecting to this computer, with the exception of those selected on the Exceptions tab.
Don't allow exceptions
Select this when you connect to public networks in less secure locations, such as airports. You will not be notified when Windows Firewall blocks programs. Selections on the Exceptions tab will be ignored.
Off (not recommended)
Avoid using this setting. Turning off Windows Firewall may make this computer more vulnerable to viruses and intruders.
What else should I know about Windows Firewall?
OK
Cancel



- Check that **Don't allow exceptions** is not selected on the **General** tab.
- Go to the **Exceptions** tab and select the `SJphone.exe` checkbox.
- Click the **Edit** and then **Change scope** buttons and check that the **Any computers (including those on the Internet)** option is selected.


Edit a Program
You can allow communication with this program from any computer, including those on the Internet or just from computers on your network.
Name: SJphone
Path: C:\Program Files\SJLabs\SJphone\SJphone.exe
Change scope...
OK
Cancel


Change Scope
To specify the set of computers for which this port or program is unblocked, click an option below.
To specify a custom list, type a list of IP addresses, subnets, or both, separated by commas.
Any computer (including those on the Internet)
My network (subnet) only
Custom list:
Example: 192.168.114.201,192.168.114.201/255.255.255.0
OK
Cancel

- Click the **OK** button.

Now **SJphone™** will be able to receive incoming calls. No other additional settings are required for Windows or **SJphone™**.

If you want to add manually SJphone™ to the list of programs that can accept incoming connections,

- Go to **Security Center** and click **Windows Firewall**.
- Check that **Don't allow exceptions** is not selected on the **General** tab.
- Go to the **Exceptions** tab and click the **Add Program** button.
- Click the **Browse** button and select `SJphone.exe` in its home folder. Default is `…\Program Files\SJphone 1.65`.
- Click the **Change scope** button, check that the **Any computers (including those on the Internet) option** is selected, and click the **OK** button.

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- Check that `SJphone.exe` is in the **Program and Services** list on the **Exceptions** tab, and its checkbox is selected,
- Click the **OK** button.

Now **SJphone™** will be able to receive incoming calls. No other additional settings are required for Windows or **SJphone™**.

IV. How to:

This chapter explains how to perform basic **SJphone™** operations.

4.1. Place a Call

You can place a call to any computer that has H.323 or SIP compatible software running, an H.323 or SIP compatible Internet phone, or a regular phone through a Gateway or a Gatekeeper.

To place a call, select an appropriate profile and type in the **Call to:** field on the **Main** panel one of the following:

- **An IP address or Computer Name for a direct PC-to-PC call**
  Depending on the VoIP protocol you want to use, you need to select either an H.323 direct or SIP direct profile. Go to the **Use Service Profiles** section for details.

  If your computer is on a private network and you need to place a direct PC-to-PC call,
  for the SIP VoIP protocol: you may place such call only to a host in the same private network. Direct SIP calls outside the private network most likely will fail.

  for the H.323 VoIP protocol: you may place such call only to a host in the same private network. Direct H.323 calls outside the private network will fail.
- **A Caller ID for a call through a SIP Proxy**
  You need to select a profile for calls through a SIP Proxy. If there are no such profiles available, you need to create such profile by yourself. Go to the **Creating Service Profiles** section for details.
  If your computer is on a private network, you need to place the call using an available NAT traversal technology. Go to the **NAT Traversal and SIP** section for details.
- **An Alias (User Name) on an H.323 Gatekeeper**
  You need to select a profile for calls through gatekeepers. If there are no such profiles available, you need to create such profile by yourself. Go to the **Creating Service Profiles** section for details.
- **A Regular Telephone Number**
  You need to select a profile for calls through a proper SIP Proxy, or H.323 Gatekeeper or Gateway if you want to call a regular telephone number. Ask the server administrator for details. If there are no such profiles available, you need to create such profile by yourself. Go to the **Creating Service Profiles** section for details.
  You may also need to add a "prefix" to the phone number to route calls to phones through required gateways. Ask the gateway administrator for details.
  If you have **Microsoft Outlook** or **Outlook Express** installed on your computer, you may call the numbers stored in its contacts. Go to the **Contacts** section for details.
  You may also use a Gatekeeper to make a call to a regular telephone number.
- **A Nickname in the Phonebook**
  You need to specify all necessary information in the **Phonebook** about the nickname you want to type in. This can be done on the **Contacts** panel.
- **An H.323 Address**
  SJphone™ supports the advanced H.323 address syntax. No service profiles are required for this call.
  H.323 Address Examples
  If you want to call
  - A host `jd.example.com (192.168.1.12)`
    Type in the **Call To:** field: `H323:IP:jd.bigcom.com` or `H323:IP:192.168.1.12`
  - A phone number `135-7975` through the H.323 Gateway `h323gateway.example.com`

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Type in the **Call To:** field: `H323:GW:h323gateway.example.com:1357975`

- A user with the nickname `doe` registered with the H.323 Gatekeeper
  Type in the **Call To:** field: `H323:GK:H323ID:doe`
- A phone number `135` registered with the H.323 Gatekeeper
  Type in the **Call To:** field: `H323:GK:E164:135`
- An external phone number `9-1357975` through the H.323 Gatekeeper
  Type in the **Call To:** field: `H323:GK:E164:91357975`

- **A SIP URI**
  SJphone™ supports the advanced SIP URI syntax. No service profiles are required for this call.
  SIP URI Examples
  If you want to call
  - A host `jd.example.com` (`192.168.1.12`)
    Type in the **Call To:** field: `sip:jd.example.com` or `sip:192.168.1.12`
  - A user `johndoe` at `jd.example.com`
    Type in the **Call To:** field: `sip:johndoe@jd.bigcom.com`
  - A phone number `135-7975` through a SIP gateway `sipgateway.example.com`
    Type in the **Call To:** field: `sip:135-7975@sipgateway.example.com;user=phone`

And click the **Dial** button on the **SJphone™ Main** panel, or press **Alt + d** or **Enter**.

Alternatively, you can use **Neighborhood Browser**, **Speed Dial** or **Contacts** to place a call.

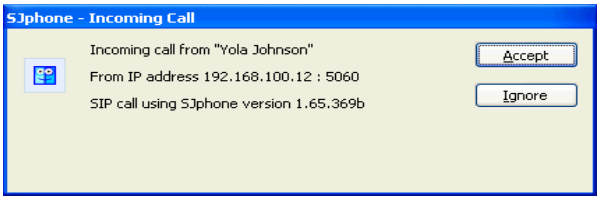
SJphone™ stores a list of recently called addresses, names, nicknames, and telephone numbers. To access this list, click the arrow next to the **Call To:** field or use the **Up** and **Down** arrow keys.

If there is no H.323 or SIP compatible software running on the computer you are calling, you will see the **Connection failed** message in the status line.

4.2. Answer a Call

You need to have **SJphone™** running to answer an incoming call.

When your computer receives a call, you will hear a ring sound. Simultaneously, an **Incoming Call** message will appear.



- **To answer the call**, click the **Accept** button, or press **Alt + a** or **Enter**.
- **To reject the call**, click the **Ignore** button, or press **Alt + i**.

You can configure your **SJphone™** to automatically accept or reject all incoming calls. Go to the **Call Options** section for details.

The **Do Not Disturb** or **Auto accept calls** modes can be also turned on and off on the shortcut menu.

You may also automatically reject all calls from anonymous respondents by selecting **Reject anonymous calls** on the shortcut menu. To access the menu, right-click the system tray icon or skin.

You may customize the sound **SJphone™** plays when a call is incoming. Go to the **Customized Incoming Call Sound** section for details.

4.3. Control Calls

You may put your active call on hold or transfer it to another respondent.

Works not for all call types. If **SJphone™** cannot control the call, the **Hold** and **Transfer** buttons are disabled.

To mute an active call,

- Click the **Mute** button
  or
  Right-click the call and select **Mute microphone** in the shortcut menu.

To un-mute a muted call

- Click the **Speak** button
  or
  Right-click the call and clear **Mute microphone** in the shortcut menu.

To put an active call on hold,

- If the conference mode is disabled, double-click a call you want to put on hold
  or
  Select a call you want to put on hold and click the **Hold** button
  or
  Right-click a call you want to put on hold and select **Hold** on the shortcut menu

To resume a call on hold,

- If the conference mode is disabled, double-click a call you want to resume
  or
  Select a call you want to resume and click the **Hold** button
  or
  Right-click a call you want to resume and select **Resume** on the shortcut menu

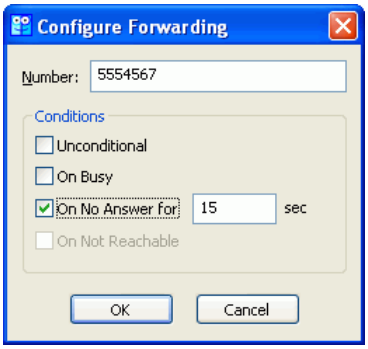
To transfer an active call to another respondent,

Blind transfer:

- Select a call you want to transfer and click the **Transfer** button
- Select **Enter number** on the shortcut menu
- Enter the respondent address in the **Call To** field and
- Click the **Transfer** button once again or press **Enter**

Attended transfer:

- Establish a call to a respondent you want to transfer call to
- Select a call you want to transfer and click the **Transfer** button
- Select the respondent on the shortcut menu



To forward all calls to another respondent,

- When there are no calls, click the **Transfer** button

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- Enter the respondent address in the **Call Forwarding** dialog box, select conditions, and click the **OK** button.

All incoming calls will be forwarded to the respondent and their records will appear in the **Missed Calls** log.

4.4. Switch between Respondents

On the **Main** panel, double-click the respondent with whom you want to talk. The other respondents will be put automatically on hold.

If Enable Conference Mode on the **Audio** tab is selected, such double-click adds/removes the respondent to/from the conference. Go to the **Start a Conference** section for details.

4.5. Start a Conference

Although **SJphone™** supports connections with multiple respondents, you may usually talk only to one respondent at a time. If you want to talk to several respondents simultaneously, you need to start a conference. You can do that by clicking the **Conference** button on the **SJphone™ Main** panel. Now you may talk to several respondents simultaneously. They also will hear you and each other.

If you want to exclude someone from the conference, select the respondent and put her/him on hold. To include the respondent back to the conference, select her/him and resume the call.

To switch back from the conference mode, click the **Conference** button on the **SJphone™ Main** panel once again.

You may also right-click the **SJphone™** system tray icon or skin and select **Conference Mode** on the shortcut menu.

4.6. Find Other SJphone™ Users on the Local Network

(available in the **Advanced Mode** interface only and may be unavailable for some service profiles)

When running, **SJphone™** constantly sends broadcast messages to the local network. Other **SJphone™**s running on computers on this network receive these messages and add the **SJphone™** to their **Neighborhood** list. You may enable or disable this feature. Go to the **Neighborhood Options** section for details.

To automatically find other SJphone™ users on your local network,

- Click the **Contacts** button and select the **Neighbors** tab.

Go to the **Neighbors** section for details.

4.7. Receive a Voice Mail

Note: Your service should support this feature.

If you have voice mail waiting, you will see a message showing how many messages are waiting.

To listen to the messages, right-click the system tray icon or skin and select **Voicemail** on the shortcut menu and follow instructions from the server.

4.8. Send a DTMF Signal

SJphone™ supports DTMF signal sending both for the SIP and H.323 protocols.

To send a DTMF signal,

- Click a digit button on the **Main** panel
  or
  Press a digit key on a keyboard.

The DTMF tone will be sent to the respondent's device and played on the host system through the playback sound device selected on the **Audio** tab.

If **SJphone™** plays back DTMF signals too loudly on the host system, you may disable local DTMF playback by selecting the **Disable local DTMF playback** option on the **Advanced Audio Settings** panel. (available in the **Advanced Mode** interface only and may be unavailable for some service profiles)

Depending on the VoIP protocol and service, signals may be sent using various methods. The method is specified in the service profile.

4.9. Use Service Profiles

SJphone™ may use predefined sets of options for specific types of calls or for calls through different Internet Telephony Service providers (ITSP). Such sets of options are stored in profiles. You may even create your own profile with your own set of options, such as your SIP server, Gatekeeper, or Gateway parameters, although that requires some knowledge of VoIP technology.

If you use several ITSPs, you may easily switch between them just selecting a required profile, usually supplied by the service provider.

Some profiles require initializing, that is, they require the user to enter some account-specific information, such as a phone number, access password, etc. A **Service:** message will appear the first time you select such profile. You may also re-initialize the profile, if its account-specific information changes.

To initialize or re-initialize a profile,

- Right-click the **SJphone™** system tray icon or skin, select **Services**, and then the required profile
- Right-click the **SJphone™** system tray icon or skin once again and select **Re-initialize selected service**.

The **Service:** message will appear

- Enter the required information and click the **OK** button

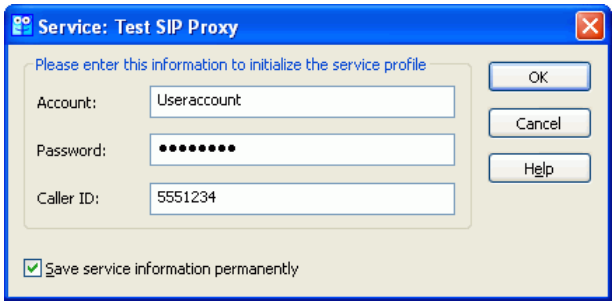
or

- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab.
- Select the required profile and click the **Initialize** button.

The **Service:** message will appear

- Enter the required information and click the **OK** button

Select Save service information permanently if you do not want to enter this information every time you select this profile.



To clear the initialization information of a profile,

- Right-click the **SJphone™** system tray icon or skin, select **Services**, and then the required profile
- Right-click the **SJphone™** system tray icon or skin once again and select **Logout**.

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To select a profile,

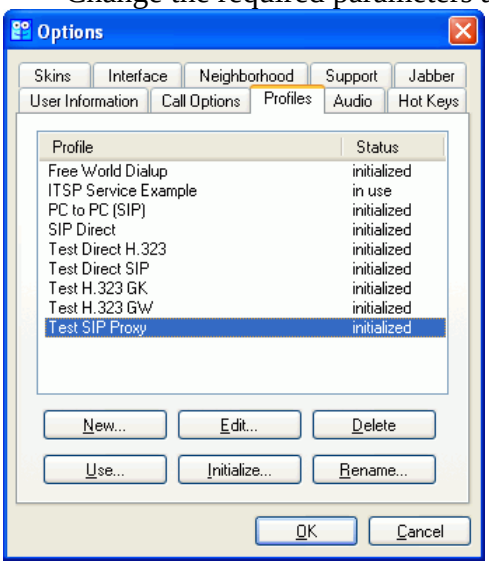
- Right-click the **SJphone™** system tray icon or skin, select **Services**, and then the required profile

or

- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab.
- Select the required profile and click the **Use** button.

To edit a profile,

- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab.
- Select the required profile and click the **Edit** button.
- Change the required parameters and click the **OK** button



New	Click this button to create a new profile
Edit	Click this button to edit a selected profile
Delete	Click this button to delete a selected profile
Use	Click this button to switch to a selected profile
Initialize	Click this button to initialize a selected profile
Rename	Click this button to rename a selected profile

To delete a profile,

- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab.
- Select the required profile and click the **Delete** button.

A **Delete Profile** message will appear.

- Click the **Yes** button to confirm profile remove.

Attention! If you accidentally remove a profile, the only way to restore the profile is to create it once again.

To rename a profile,

- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab.
- Select the required profile and click the **Rename** button.

A **Rename Profile** message will appear.

- Enter a new profile name and click the **OK** button.

4.10. Create a Desktop Shortcut to a Respondent

You may create a desktop shortcut to a respondent. When it is created you may call this respondent simply by clicking this desktop shortcut. You may specify either a callto URL or a SIP URI.

To create a desktop shortcut,

- Right-click anywhere on the desktop and select **New -> Shortcut**

The **Create Shortcut** wizard will appear.

- Specify either

A **SIP URI** to the respondent. The **SIP URI** is described in the **SIP URI Syntax** section. Please note if **SJphone™** is registered with a strictly outbound SIP proxy, the **SIP URI** will be processed by the proxy and it may reject a call if the **SIP URI** is outside proxy's domains.
If you want to call

- A host `jd.example.com`
  Type in the **Create Shortcut** wizard `sip:jd.example.com`
- A host with the IP address `192.168.12.12`
  Type in the **Create Shortcut** wizard `sip:192.168.12.12`
- A phone number `555-1234` through a SIP proxy `proxy.example.com`
  Type in the **Create Shortcut** wizard `sip:5551234@proxy.example.com`

A **callto URL** to the respondent. The **callto URL** is described in the **Command Line Switches** section.
If you want to call

- A host `jd.example.com`
  Type in the **Create Shortcut** wizard `callto:jd.example.com`
- A host with the IP address `192.168.12.12` using the H.323 VoIP protocol
  Type in the **Create Shortcut** wizard `callto:192.168.12.12+type=ip`
- A host with the IP address `192.168.12.12` using the SIP VoIP protocol
  Type in the **Create Shortcut** wizard `callto:192.168.12.12+type=sip`

4.11. End a Call

To end a call,

- Select the respondent and click the **Hangup** button on the **SJphone™ Main** panel, or press **Alt + h**,
  or
  Right-click the respondent and select **Hangup** on the shortcut menu.

You may end several calls simultaneously. Select the respondents, right-click them, and select **Hangup** on the shortcut menu.

4.12. Restart SJphone™

To restart SJphone™,

- Right-click the **SJphone™** system tray icon or skin and select **Restart** on the shortcut menu.

4.13. Exit SJphone™

To exit SJphone™,

- Right-click the **SJphone™** system tray icon or skin and select **Shut down** on the shortcut menu.
  or
  Click the **Exit** button on the **Main** panel (without skin).

Note: When you exit **SJphone™** you will be unable to receive incoming calls.

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V. Advanced Features

The following features are for advanced users. You need to be familiar with the details of the ITU-T H.323 and SIP protocols, audio codecs, and other technical details in order to use them correctly.

5.1. Creating Profiles

(may be unavailable for some service profiles)

SJphone™ may use predefined sets of options for specific types of calls or for calls through different Internet Telephony Service providers (ITSP). Such sets of options are stored in profiles.

An example of a call-type specific profile is that for the direct call to a host using the SIP VoIP protocol. You do not need to enter a complex dial string like `sip:johndoe@jd.example.com`. Instead, you may select the SIP direct profile and type the URL in the **Call to:** field. Or, if you use several ITSPs, you may easily switch between them just selecting a required profile, usually supplied by the service provider.

You may even write your own profile with your own set of options, such as your SIP Proxy, Gatekeeper, or Gateway parameters, although that requires some knowledge of VoIP technology.

Some profiles require initializing, that is, they require the user to enter some account-specific information, such as a phone number, access password, etc. You will enter this information during profile initialization. Initialization information may be either permanently stored, or you will have to enter it every time you switch to this profile. You will specify on the **Initialization** tab which information is required for initialization and how it will be stored.

Initialization options:

Inquired	If selected, this parameter will be asked during profile initialization
Saved	If selected, this parameter will be permanently saved
Required	If selected, this parameter is required for this profile

Attention! You should have enough privileges in the system to create a profile. (For example, you should be in the Administrators or Power users user group rather than the Users group.)

SIP Profiles

Calls through a SIP Proxy

Note: If your **SJphone™** is behind NAT, please read the **NAT traversal and SIP** section.

This profile is used to make calls to another computer using a registration with a SIP proxy.

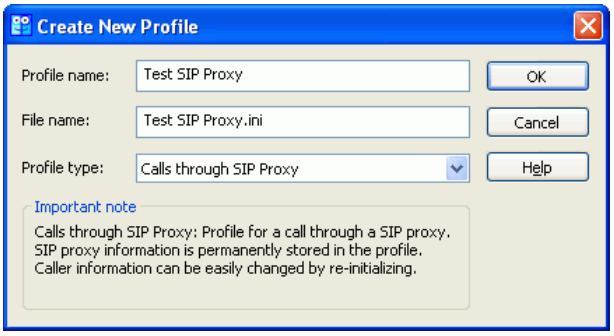
To create a new profile for calls through a SIP proxy,

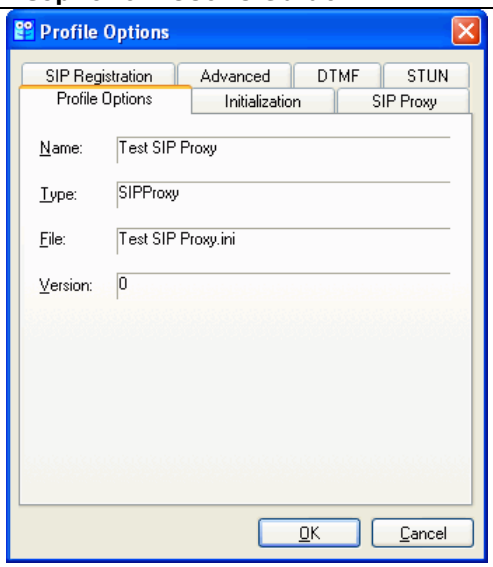
- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab
- Click the **New** button

A **Create New Profile** window will appear

- Select Calls through SIP Proxy in Profile, specify a name for the profile, and click the **OK** button.
- Specify the required profile options and click the **OK** button.

The **Profile options** tab will show the name, type, and other profile's properties.





Initialization

Accounts	Your account (login) at the proxy
Password	Your password at the proxy
Caller ID	Your Caller ID at the proxy. Used to form your SIP Address of Record
Full Address of Record	Your Full Address of Record. Used to form your SIP Address of Record

Either Caller ID or Full Address of Record can be specified.

SJphone™ forms your SIP Address of Record in the following way:

If Full Address of Record is specified explicitly, it is used as the SIP Address of Record regardless of other parameters.

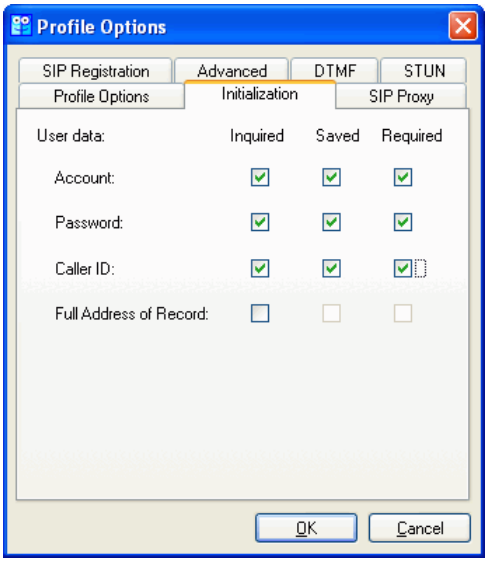
Otherwise, the Address of Record is a string of the following form:

```
sip:CallerID@UserDomain
```

Where

CallerID is selected on the **Initialization** tab and specified in the Caller ID option during initialization. If this option is not selected, Account will be used.

UserDomain is specified in the Domain: option on the **SIP Proxy** tab.



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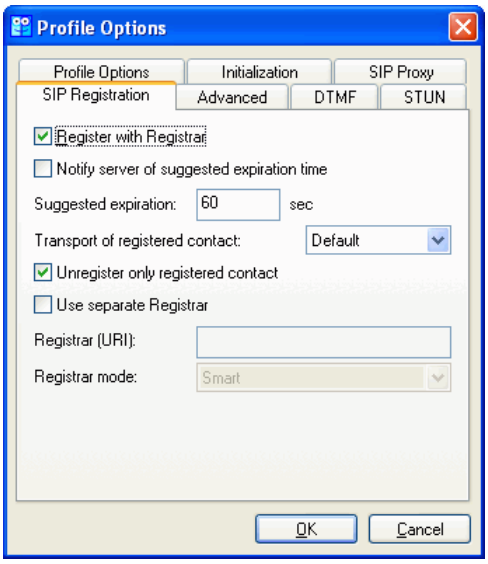
SIP Proxy



Domain/Realm

The domain part of the full Address of Record if it is not specified explicitly during profile initialization. It is also used to form all destination URIs when the user dials a telephone number/user name without specifying its domain explicitly. May be either a DNS name or IP address, and a port may be specified. If an ITSP or system administrator does not provide you with this parameter, most likely it coincides with Proxy URI:

Examples:

```
example.com
example.com:5090.
```

Use Outbound Proxy

Usually this option should be selected, except when the proxy is a redirect server.

If this option is selected, **SJphone™** will initiate all its SIP communication through the outbound proxy.

If this option is cleared, **SJphone™** will send all its SIP requests directly to the destination SIP URI using standard URI DNS resolving procedures (RFC 3263).

Recommended status: Selected.

Proxy (URI):

A full SIP URI of the Outbound proxy. In the simplest case, it may be either a DNS name or IP address of the proxy, and a port. Should start with the `sip:` prefix. Generally, it may contain the URI parameters from the SIP protocol specifications.

Examples:

```
sip:proxy.example.com
sip:proxy.example.com:5060
sip:192.0.2.124:5678;transport=tcp;lr
```

Proxy usage mode: This option sets the way **SJphone™** communicates with the proxy. Do not change this option unless instructed otherwise.

There are 3 modes available:

Smart: **SJphone™** initiates a new SIP communication through the proxy and continues it according to the dialog route set. This mode enables **SJphone™** to receive direct SIP calls, too.

Pre-existing route set: **SJphone™** communicates with the proxy using the pre-existing route set as described in RFC 3261. Note: Although RFC 3261 recommends this mode instead of the strict outbound proxy mode, but some SIP servers support this mode incorrectly.

Strict outbound proxy: **SJphone™** sends all SIP requests and responses through the proxy. In this mode, **SJphone™** cannot receive direct SIP calls.

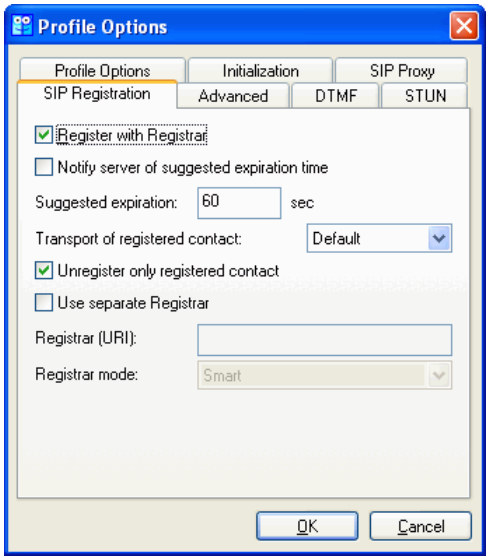
Recommended status: Smart.

Use separate Outbound Proxy for NAT: If this option is selected, **SJphone™** will use a separate proxy when it has discovered that it is behind NAT.

NAT Proxy (URI): A full SIP URI of the proxy used instead of Proxy URI: when **SJphone™** is behind NAT and Use separate Outbound Proxy for NAT is selected.

NAT Proxy mode: This option sets the way **SJphone™** communicates with the NAT proxy.

SIP Registration



Register with Registrar: If this option is selected, **SJphone™** registers with the SIP server. Always select this option unless instructed otherwise.

Recommended status: Selected.

Notify server of default expiration time: Suggested expiration interval that indicates how long **SJphone™** would like the registration to be valid.

Transport of registered contact: This option specifies the suggested signalling transport protocol to be used for incoming calls. Do not change this option unless instructed otherwise.

Default value: Default (unspecified).

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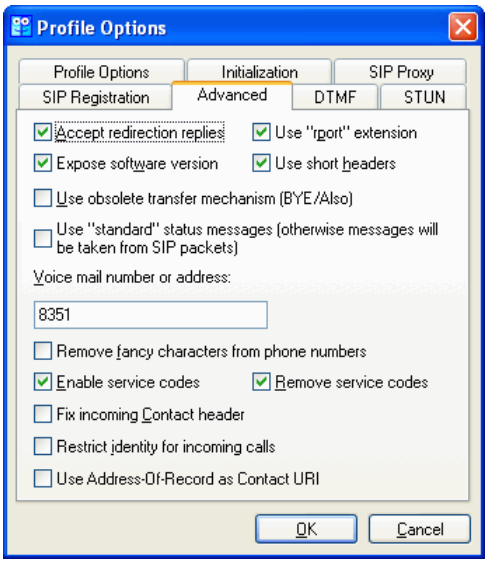
Unregister only registered contact	If this option is selected, during the process of unregistering, **SJphone™** will unregister only its temporary SIP URI associated with the current running **SJphone™** instance. If this option is cleared, **SJphone™** will unregister all temporary SIP URIs associated with the permanent user address-of-record. Recommended status: Selected.
Use separate Registrar	A very rarely used option. If this option is selected, **SJphone™** will register with a separate SIP entity rather than with the proxy specified in the Proxy URI option. Recommended status: Cleared.
Registrar (URI):	A full SIP URI of the Registrar server if the Use separate Registrar option is selected.
Registrar mode:	This option sets the way **SJphone™** communicates with the separate Registrar server.

Advanced



Accept redirection replies	If this option is selected, **SJphone™** will automatically call the addresses returned with redirection replies. (Valid only for SIP URI). Recommended status: Selected.
Use short headers	If this option is selected, **SJphone™** will use short header names in SIP messages. Clear this option only if your system does not support short headers. Recommended status: Selected.
Expose software version	If this option is selected, **SJphone™** will send information about itself in the `Server` and `User-Agent` headers. Recommended status: Selected.
Use obsolete transfer mechanism (BYE/Also)	If this option is selected, **SJphone™** will use the BYE/Also mechanism for blind transfers. In fact, this mechanism is deprecated. Recommended status: Cleared.

Use "standard" status messages	If this option is selected, **SJphone™** will display SIP status messages from its internal SIP message list. If the option is clear, SIP messages will be taken from incoming SIP packets. Select this option if you have a completely localized **SJphone™** version. Recommended status: Cleared.
Voice mail number or address	If the SIP proxy supports this feature, enter a voicemail number or address.
Remove fancy characters from phone numbers	If this option is selected, **SJphone™** will remove +, (,), -, and space from dialed phone numbers. Recommended status: Cleared.
Enable service codes	If this option is selected, **SJphone™** will follow issued **vertical service codes**.
Remove service codes	If this option is selected, **SJphone™** will remove **vertical service codes** before sending an entered number to the server.
Fix incoming Contact header	If this option is selected, **SJphone™** will change the SIP `Contact` header trying to compensate for packet address changes made by various NAT systems. Select it if you have problems in making calls between open Internet and private networks. Recommended status: Cleared.
Restrict identity for incoming calls	If this option is selected, **SJphone™** will conceal your name, username, and other sensitive information for incoming calls as much as possible. Please note that the policy of accepting/delivering anonymous calls varies for service providers.
Use Address-Of-Record as Contact URI	If this option is selected, **SJphone™** will insert into the `Contact` header fields of all SIP-requests and responses within a dialog its own `Address-of-Record` rather than the IP-address of its host. This may be useful for walled-garden environments in cases when a proxy is lazy to modify the `Contact` header field or/and insert the `Record-Route` header field, or it is not desirable/achievable anyway. Always have this option cleared unless instructed otherwise. Recommended status: Cleared.

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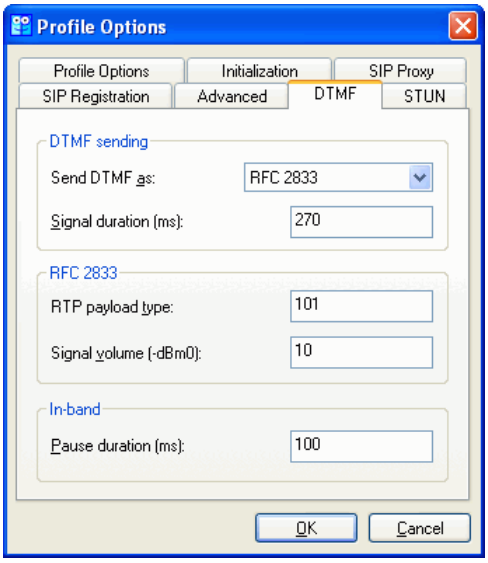
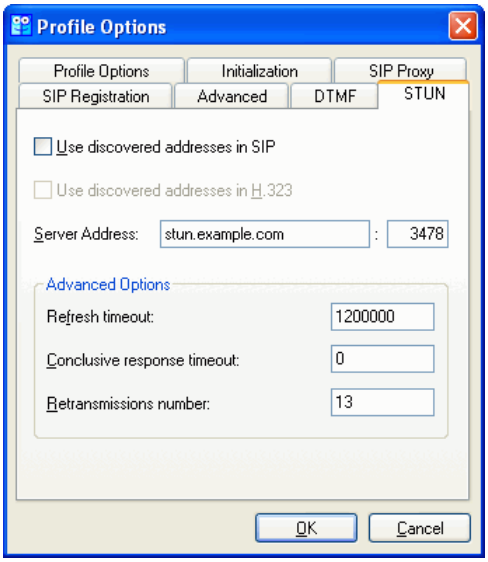
DTMF

DTMF sending methods are described in the **DTMF Signal Sending** section.



Depending on your local DTMF standard, you may need to adjust some DTMF parameters. If necessary, consult your ITSP support or/and network administrator.

STUN



Use discovered addresses in SIP	If this option is selected, **SJphone™** will use addresses discovered by the STUN procedure in SIP packets. Recommended status: Cleared.
Server Address	The address and port of the STUN server. May be either a DNS name or IP address. Unless instructed otherwise, do not change this option.
Advanced Options	Use this option if the STUN discovery procedure works unstably or does not work at all. Always consult your ITSP support or/and network administrator when in doubt. Do not change them unless you know what you are doing.
Refresh timeout	A time interval in milliseconds after which a status of each network interface is determined. Should be at least 100 msec.

Conclusive response timeout	A time interval in milliseconds during which the STUN client waits for re-transmitted answers after the first answer to the STUN query has been received. Should be at least 100 msec. RFC3489 recommends 10000. The special value 0 means that the STUN client will not wait for the re-sent answers.
Retransmission number	The maximum number of re-transmitted first STUN queries when the network interface status is being determined. Should be less than, or equal to, 13.

Initialization:

To initialize or re-initialize this profile,

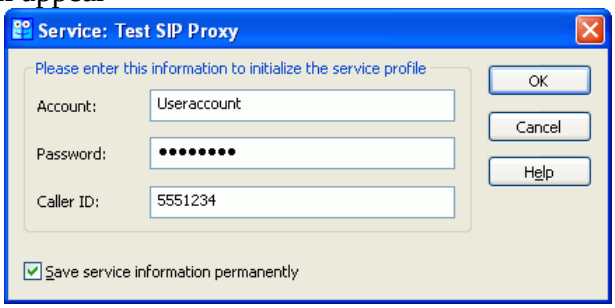
- Right-click the **SJphone™** system tray icon or skin, select **Services**, and then the required profile
- Right-click the **SJphone™** system tray icon or skin once again and select **Re-initialize selected service**

or

- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab

Select the required profile and click the Initialize button

The **Service** message will appear



- Enter the required information and click the **OK** button

Select Save service information permanently if you do not want to enter this information every time you select this profile.

Direct SIP PC-to-PC call

This profile is used to make direct calls to another computer using the SIP VoIP protocol.

To create a new profile for direct SIP PC-to-PC calls,

- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab
- Click the **New** button

A **Create New Profile** window will appear

- Select Direct SIP Calls in Profile, specify a name for the profile, and click the **OK** button.

Specify the required profile options and click the **OK** button

The **Profile Optionsl** tab will show the name, type, and other profile's properties.

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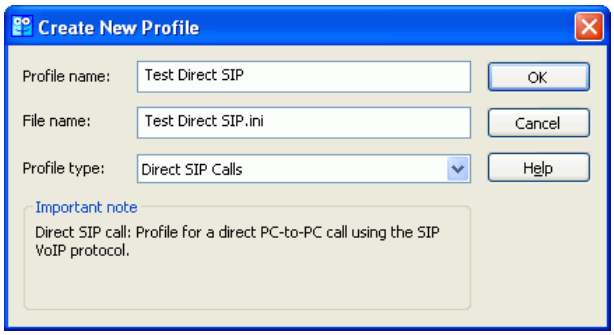
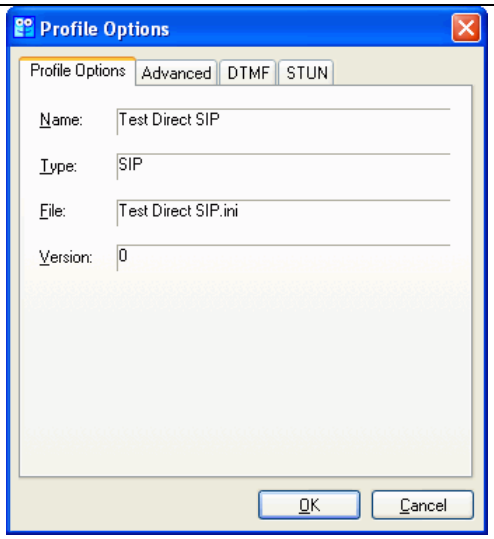
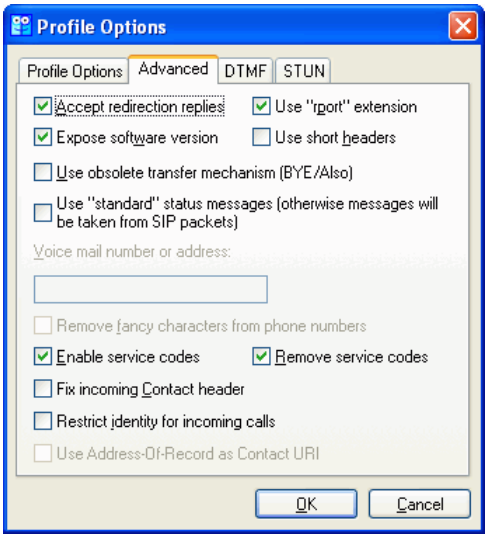




Advanced

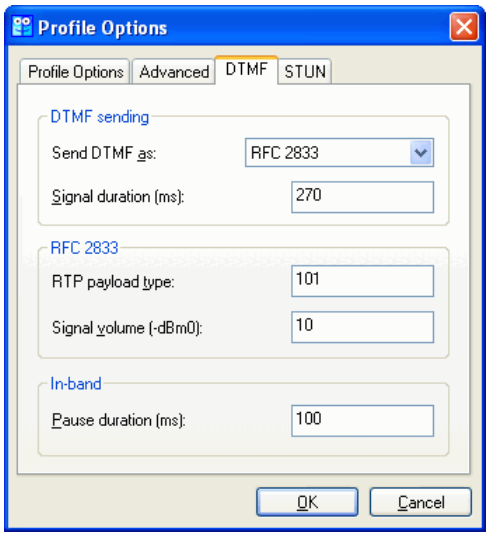


Accept redirection replies	If this option is selected, **SJphone™** will automatically call the addresses returned with redirection replies. (Valid only for SIP URI). Recommended status: Selected.
Use short headers	If this option is selected, **SJphone™** will use short header names in SIP messages. Clear this option only if your system does not support short headers. Recommended status: Selected.
Expose software version	If this option is selected, **SJphone™** will send information about itself in the `Server` and `User-Agent` headers. Recommended status: Selected.
Use obsolete transfer mechanism (BYE/Also)	If this option is selected, **SJphone™** will use the BYE/Also mechanism for blind transfers. In fact, this mechanism is deprecated. Recommended status: Cleared.

Use "standard" status messages	If this option is selected, **SJphone™** will display SIP status messages from its internal SIP message list. If the option is clear, SIP messages will be taken from incoming SIP packets. Select this option if you have a completely localized **SJphone™** version. Recommended status: Cleared.
Enable service codes	If this option is selected, **SJphone™** will follow issued **vertical service codes**.
Remove service codes	If this option is selected, **SJphone™** will remove **vertical service codes** before sending an entered number to the server.
Fix incoming Contact header	If this option is selected, **SJphone™** will change the SIP `Contact` header trying to compensate for packet address changes made by various NAT systems. Select it if you have problems in making calls between open Internet and private networks. Recommended status: Cleared.
Restrict identity for incoming calls	If this option is selected, **SJphone™** will conceal your name, username, and other sensitive information for incoming calls as much as possible. Please note that the policy of accepting/delivering anonymous calls varies for service providers.

DTMF

DTMF sending methods are described in the **DTMF Signal Sending** section.



Depending on your local DTMF standard, you may need to adjust some DTMF parameters. If necessary, consult your ITSP support or/and network administrator.

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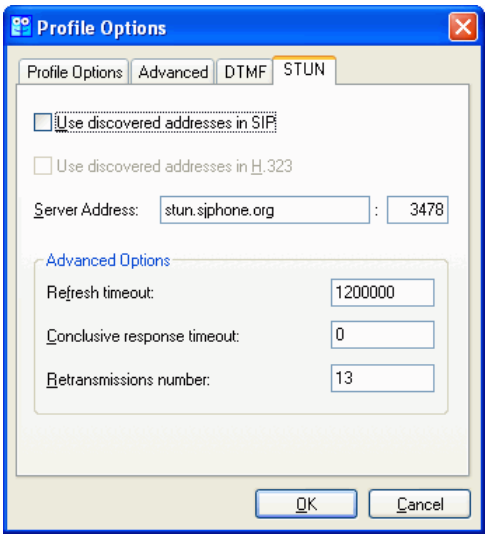
STUN



Use discovered addresses in SIP	If this option is selected, **SJphone™** will use addresses discovered by the STUN procedure in SIP packets. Recommended status: Cleared.
Server Address	The address and port of the STUN server. May be either a DNS name or IP address. Unless instructed otherwise, do not change this option.
Advanced Options	Use this option if the STUN discovery procedure works unstably or does not work at all. Always consult your ITSP support or/and network administrator when in doubt. Do not change them unless you know what you are doing.
Refresh timeout	A time interval in milliseconds after which a status of each network interface is determined. Should be at least 100 msec.
Conclusive response timeout	A time interval in milliseconds during which the STUN client waits for re-transmitted answers after the first answer to the STUN query has been received. Should be at least 100 msec. RFC3489 recommends 10000. The special value 0 means that the STUN client will not wait for the re-sent answers.
Retransmission number	The maximum number of re-transmitted first STUN queries when the network interface status is being determined. Should be less than, or equal to, 13.

Initialization: This profile does not require initialization.

H.323 Profiles

Direct H.323 PC-to-PC calls

This profile is used to make direct calls to another computer using the H.323 VoIP protocol.

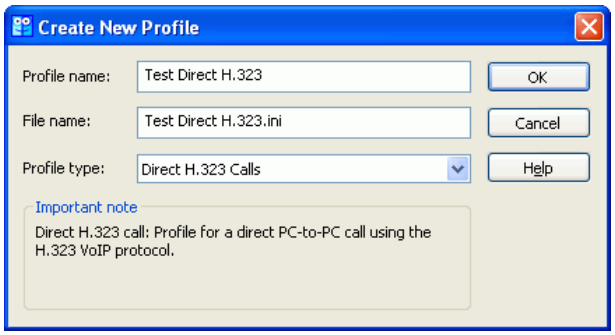
To create a new profile for direct H.323 PC-to-PC calls,

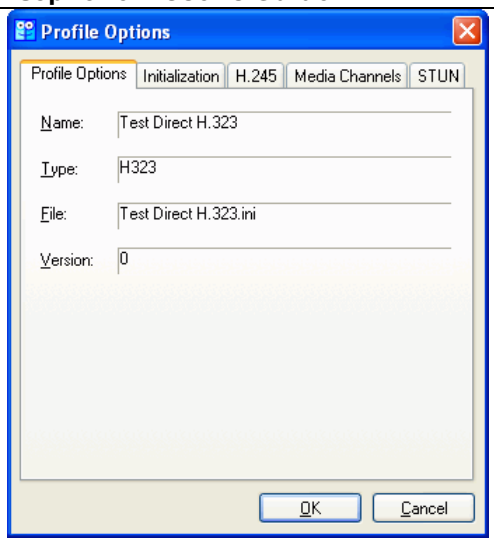
- Click the **Options** button on the **SJphone™** Main panel and go to the **Profiles** tab
- Click the **New** button

A **Create New Profile** window will appear

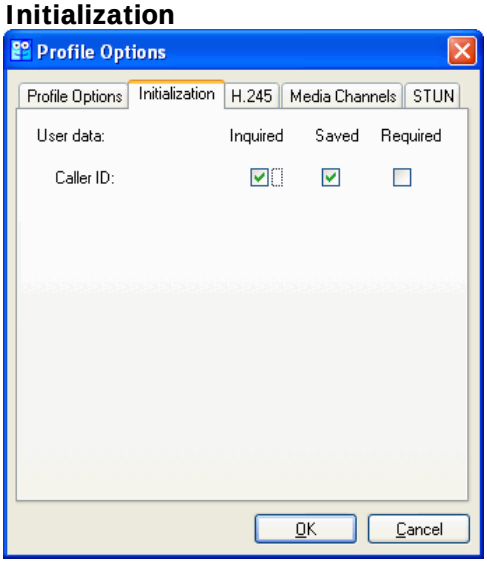
- Select Direct H.323 Calls in Profile, specify a name for the profile, and click the **OK** button.
- Specify the required profile options and click the **OK** button

The **Profile Options** tab will show the name, type, and other profile's properties.





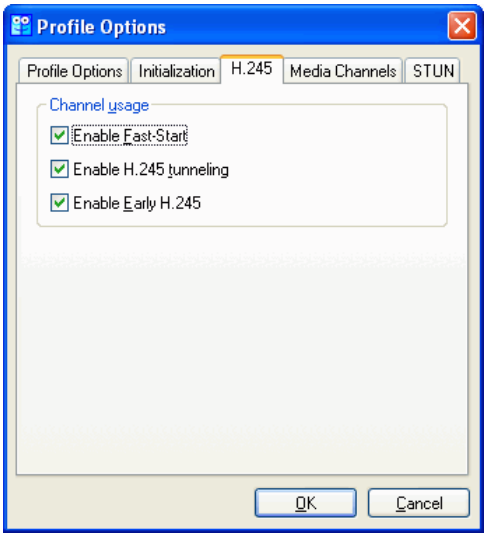
Initialization



Caller ID

This number may be used by internet or telephone network providers for call identification and billing. Contact your system administrator or VoIP service provider for details

H.245



Channel usage

These parameters are described in ITU H.323 and H.245 protocol specifications. Do not change them unless instructed by your system administrator.

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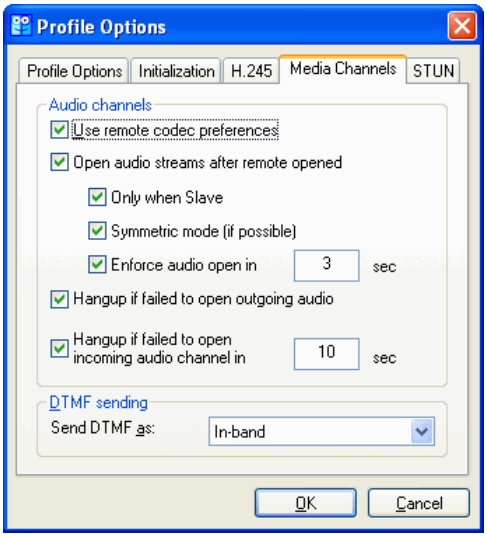
Media Channels

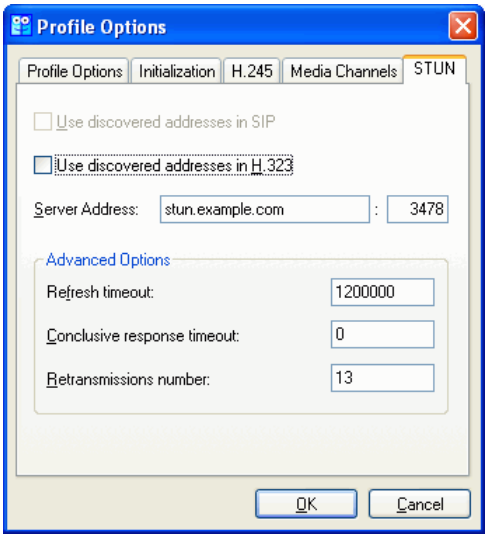
DTMF sending methods are described in the **DTMF Signal Sending** section.



Audio channels

These parameters are described in ITU H.323 and H.245 protocol specifications. Do not change them unless instructed by your system administrator.

STUN



Use discovered addresses in H.323

If this option is selected, **SJphone™** will use addresses discovered by the STUN procedure in H.323 packets. Recommended status: Cleared.

Server Address

The address and port of the STUN server. May be either a DNS name or IP address. Unless instructed otherwise, do not change this option.

Advanced Options

Use this option if the STUN discovery procedure works unstably or does not work at all. Always consult your ITSP support or/and network administrator when in doubt. Do not change them unless you know what you are doing.

Refresh timeout

A time interval in milliseconds after which a status of each network interface is determined. Should be at least 100 msec.

Conclusive response timeout	A time interval in milliseconds during which the STUN client waits for re-transmitted answers after the first answer to the STUN query has been received. Should be at least 100 msec. RFC3489 recommends 10000. The special value 0 means that the STUN client will not wait for the re-sent answers.
Retransmission number	The maximum number of re-transmitted first STUN queries when the network interface status is being determined. Should be less than, or equal to, 13.

Initialization:

To initialize or re-initialize this profile,

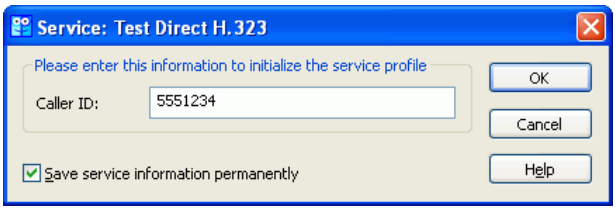
- Right-click the **SJphone™** system tray icon or skin, select **Services**, and then the required profile
- Right-click the **SJphone™** system tray icon or skin once again and select **Re-initialize selected service**

or

- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab

Select the required profile and click the Initialize button

The **Service** message will appear



- Enter the required information and click the **OK** button

Select Save service information permanently if you do not want to enter this information every time you select this profile.

Calls through an H.323 Gateway

This profile is used to make calls to PSTN networks through an H.323 Gateway.

To create a new profile for calls through an H.323 gateway,

Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab

- Click the **New** button

A **Create New Profile** window will appear

- Select Calls through H.323 Gateway in Profile, specify a name for the profile, and click the **OK** button.
- Specify the required profile options and click the **OK** button

The **Profile Options** tab will show the name, type, and other profile's properties.

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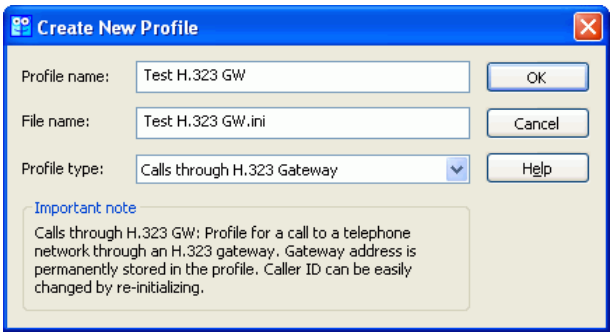
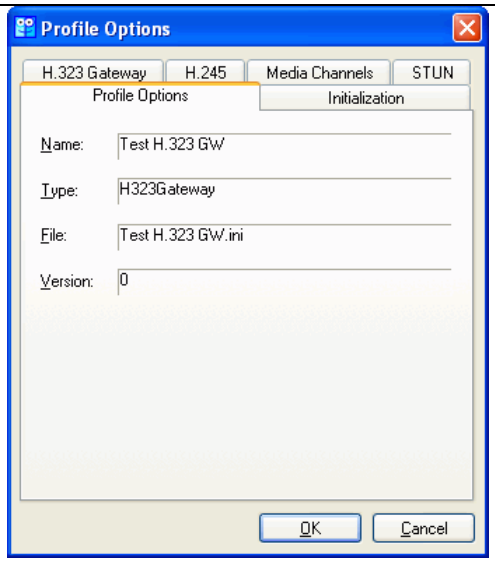
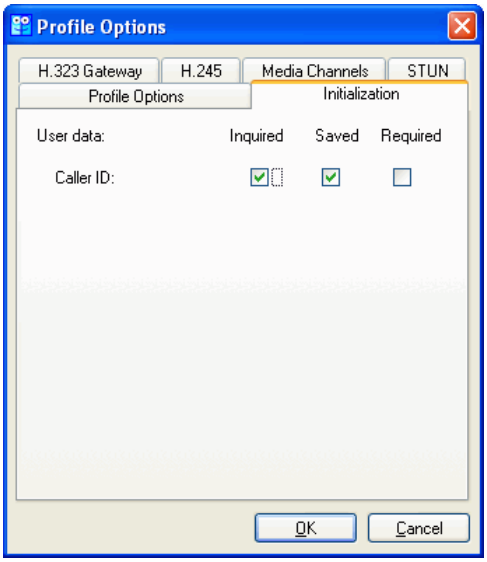

Create New Profile
Profile name:
Test H.323 GW
File name:
Test H.323 GW.ini
Profile type:
Calls through H.323 Gateway
OK
Cancel
Help
Important note
Calls through H.323 GW: Profile for a call to a telephone network through an H.323 gateway. Gateway address is permanently stored in the profile. Caller ID can be easily changed by re-initializing.



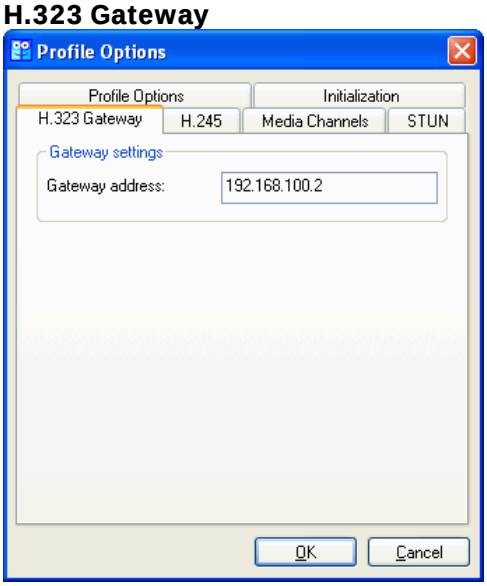
Initialization



Caller ID

This number may be used by the gateway for call identification and billing. Contact your system administrator or VoIP service provider for details

H.323 Gateway

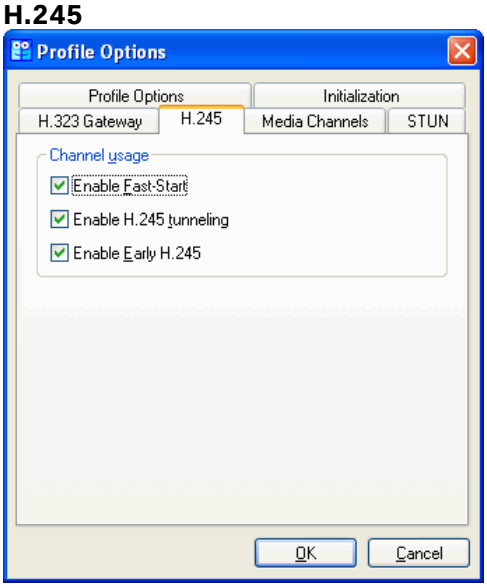


Gateway settings

Gateway address

The Gateway IP address

H.245

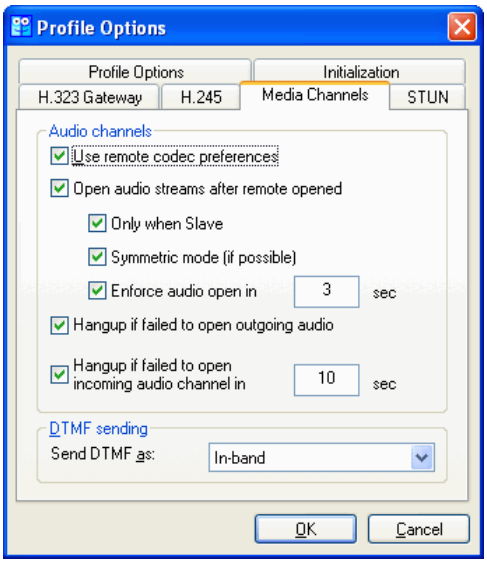


Channel usage

These parameters are described in ITU H.323 and H.245 protocol specifications. Do not change them unless instructed by the gateway administrator.

Media Channels

DTMF sending methods are described in the **DTMF Signal Sending** section.



Audio channels

These parameters are described in ITU H.323 and H.245 protocol specifications. Do not change them unless instructed by the gateway administrator.

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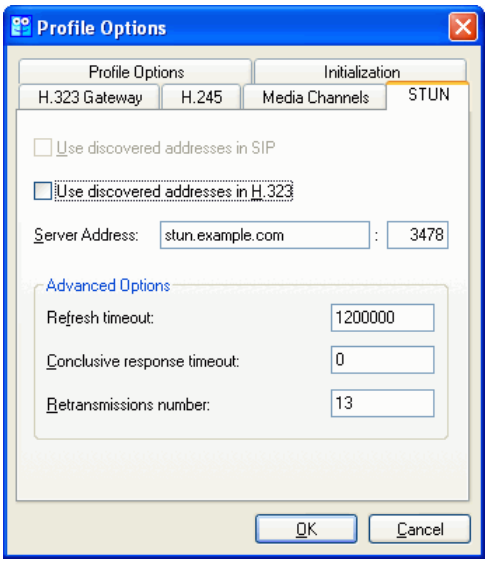
STUN



Use discovered addresses in H.323	If this option is selected, **SJphone™** will use addresses discovered by the STUN procedure in H.323 packets. Recommended status: Cleared.
Server Address	The address and port of the STUN server. May be either a DNS name or IP address. Unless instructed otherwise, do not change this option.
Advanced Options	Use this option if the STUN discovery procedure works unstably or does not work at all. Always consult your ITSP support or/and network administrator when in doubt. Do not change them unless you know what you are doing.
Refresh timeout	A time interval in milliseconds after which a status of each network interface is determined. Should be at least 100 msec.
Conclusive response timeout	A time interval in milliseconds during which the STUN client waits for re-transmitted answers after the first answer to the STUN query has been received. Should be at least 100 msec. RFC3489 recommends 10000. The special value 0 means that the STUN client will not wait for the re-sent answers.
Retransmission number	The maximum number of re-transmitted first STUN queries when the network interface status is being determined. Should be less than, or equal to, 13.

Initialization:

To initialize or re-initialize this profile,

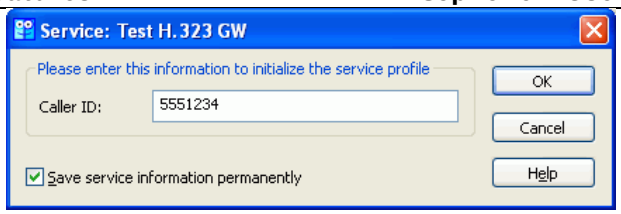
- Right-click the **SJphone™** system tray icon or skin, select **Services**, and then the required profile
- Right-click the **SJphone™** system tray icon or skin once again and select **Re-initialize selected service**

or

- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab

Select the required profile and click the Initialize button

The **Service** message will appear



- Enter the required information and click the **OK** button

Select Save service information permanently if you do not want to enter this information every time you select this profile.

Calls through an H.323 Gatekeeper

This profile is used to make calls through an H.323 Gatekeeper.

When you place a call to a regular phone through a gatekeeper, you may need to add a "prefix" (a special digit) to the phone number. Such prefixes are used to route calls to phones through required gateways. Ask the gatekeeper administrator for details.

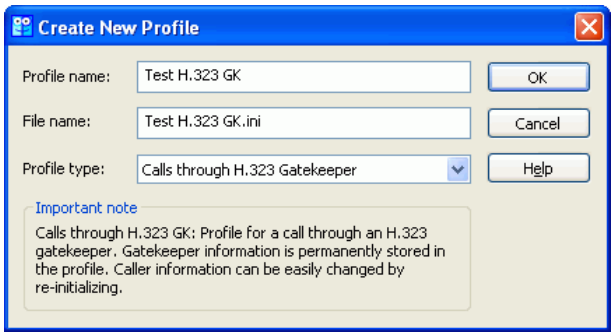
To create a new profile for calls through an H.323 gatekeeper,

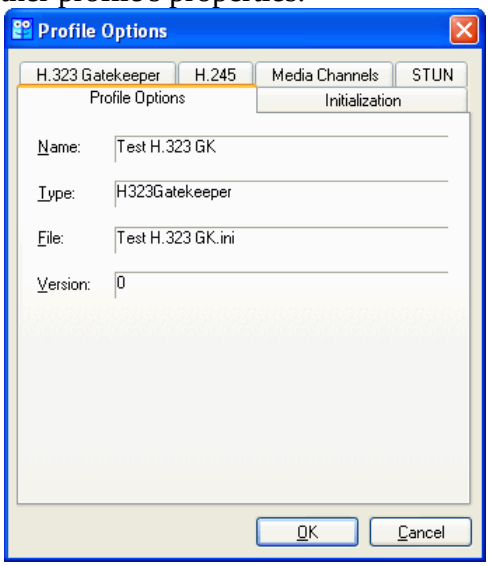
- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab
- Click the **New** button

A **Create New Profile** window will appear

- Select Calls through H.323 Gatekeeper in Profile, specify a name for the profile, and click the **OK** button.
- Specify the required profile options and click the **OK** button

The **Profile Options** tab will show the name, type, and other profile's properties.


Create New Profile
Profile name:
Test H.323 GK
File name:
Test H.323 GK.ini
Profile type:
Calls through H.323 Gatekeeper
OK
Cancel
Help
Important note
Calls through H.323 GK: Profile for a call through an H.323 gatekeeper. Gatekeeper information is permanently stored in the profile. Caller information can be easily changed by re-initializing.



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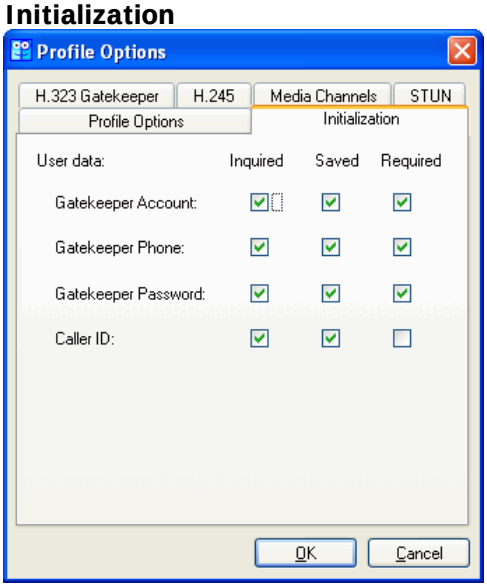
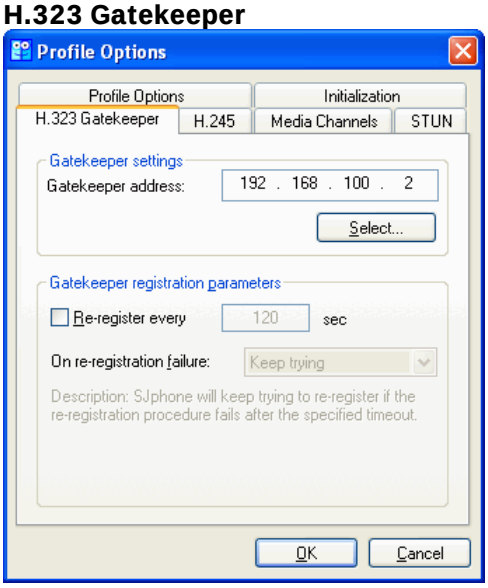
Initialization



Gatekeeper Account	A user account on the gatekeeper
Gatekeeper Phone	You may register with a gatekeeper using only a phone number. If someone calls this number, the gatekeeper will direct the call to you. Contact the gatekeeper administrator for details.
Gatekeeper Password	A user password on the gatekeeper
Caller ID	This number may be used by the gatekeeper for call identification and billing. Contact your gatekeeper administrator or VoIP service provider for details

H.323 Gatekeeper



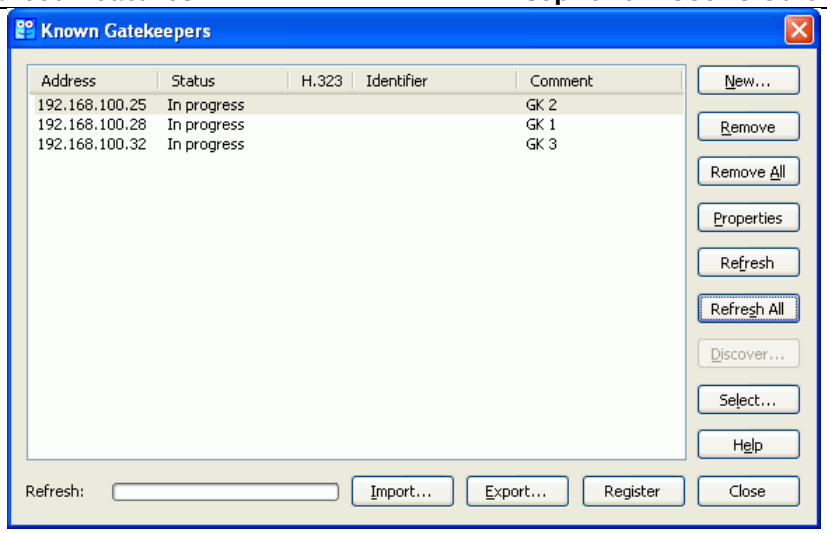
Gatekeeper settings	The Gatekeeper IP address.
Gatekeeper address	
Gatekeeper registration parameters	
Reregister every...seconds	Some gatekeepers require their users to re-register after a certain time interval. Time interval between re-registration with a gatekeeper. If you want to disable this feature, clear this option.
On re-registration failure	This option specifies what **SJphone™** will do if it fails to re-register with a gatekeeper. Available options are Keep trying to re-register, or Continue unregistered. In the latter case, it will stop attempts to re-register, alert the user, and disable gatekeeper usage.

List of known gatekeepers

Your system administrator or VoIP service provider may give you a file with a list of known gatekeepers. Such files can be used for **SJphone™** versions for all OS

Click the **Select Gatekeeper** button to activate the **Known Gatekeepers** panel.

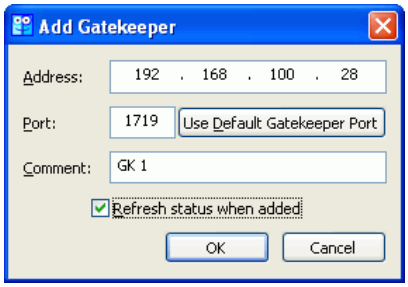
Address	Shows the gatekeeper address
Status	Shows the gatekeeper status. The status may be: Indeterminate, In progress, Available, Rejected, Not responding
H.323	Shows the version of the H.323 protocol that the gatekeeper supports
Identifier	Shows the gatekeeper identifier
Comment	Shows gatekeeper comments.
Refresh	Shows refresh progress



Buttons

New	Click this button to add a new gatekeeper to the list
Remove	Click/ this button to remove the selected gatekeeper
Properties	Click this button to see the properties of the selected gatekeeper
Select	Click/ this button to select all gatekeepers with the same status
Close	Click/ this button to close the list of known gatekeepers
Import	Click this button to import a file with a list of gatekeepers. The default file extension is `*.gkl`.
Export	Click this button to export the list of gatekeepers into a file. The default file extension is `*.gkl`.
Remove All	Click this button to remove all gatekeepers from the list
Refresh	Click this button to refresh the status of the selected gatekeeper
Refresh All	Click this button to refresh the status of all gatekeepers in the list
Register	Click this button to register with the selected gatekeeper
Discover	Not implemented yet

You may add gatekeepers to the list. Click the **New** button and an **Add Gatekeeper** window will appear. Enter the required information about the gatekeeper and click the **OK** button.



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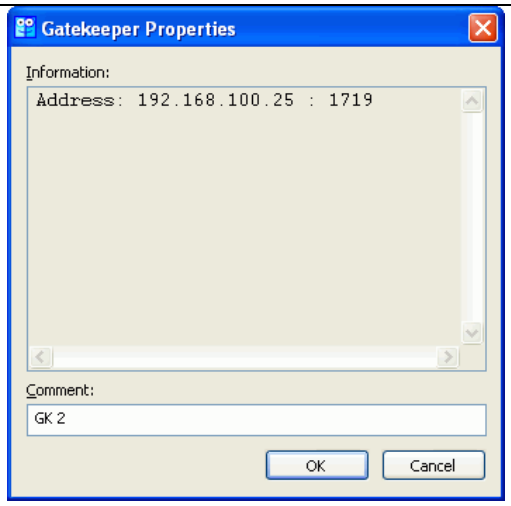
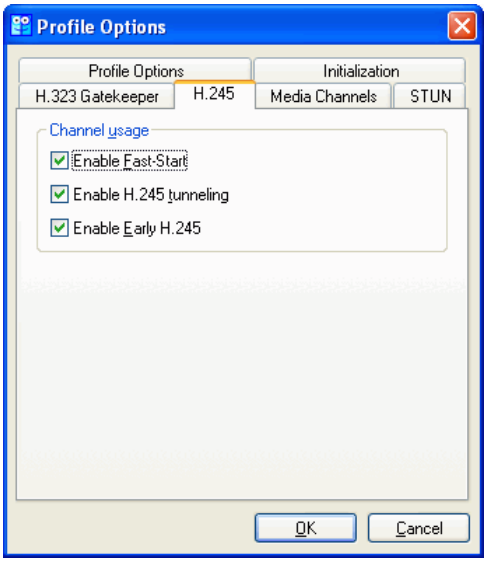
You may see the properties of each gatekeeper. Click the **Properties** button and the **Gatekeeper Properties** window will appear.



H.245

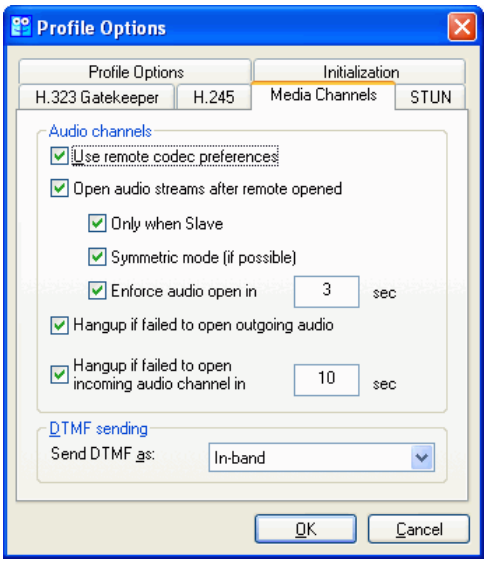


Channel usage

These parameters are described in ITU H.323 and H.245 protocol specifications. Do not change them unless instructed by the gateway administrator.

Media Channels

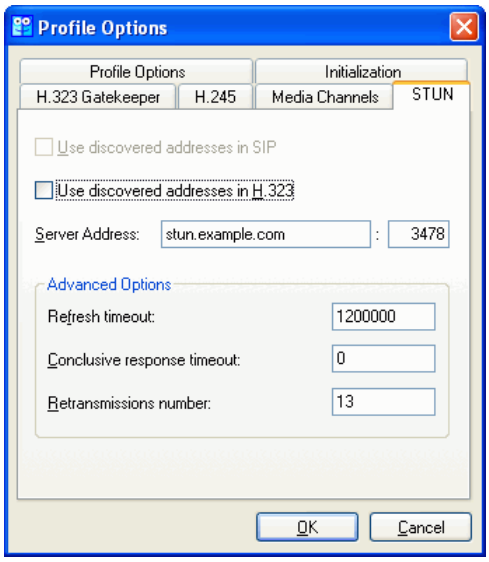
DTMF sending methods are described in the **DTMF Signal Sending** section.



Audio channels

These parameters are described in ITU H.323 and H.245 protocol specifications. Do not change them unless instructed by the gateway administrator.

STUN



Use discovered addresses in H.323	If this option is selected, **SJphone™** will use addresses discovered by the STUN procedure in H.323 packets. Recommended status: Cleared.
Server Address	The address and port of the STUN server. May be either a DNS name or IP address. Unless instructed otherwise, do not change this option.
Advanced Options	Use this option if the STUN discovery procedure works unstably or does not work at all. Always consult your ITSP support or/and network administrator when in doubt. Do not change them unless you know what you are doing.
Refresh timeout	A time interval in milliseconds after which a status of each network interface is determined. Should be at least 100 msec.
Conclusive response timeout	A time interval in milliseconds during which the STUN client waits for re-transmitted answers after the first answer to the STUN query has been received. Should be at least 100 msec. RFC3489 recommends 10000. The special value 0 means that the STUN client will not wait for the re-sent answers.
Retransmission number	The maximum number of re-transmitted first STUN queries when the network interface status is being determined. Should be less than, or equal to, 13.

Initialization:

To initialize or re-initialize this profile,

- Right-click the **SJphone™** system tray icon or skin, select **Services**, and then the required profile
- Right-click the **SJphone™** system tray icon or skin once again and select **Re-initialize selected service**

or

- Click the **Options** button on the **SJphone™ Main** panel and go to the **Profiles** tab

Select the required profile and click the Initialize button

The **Service** message will appear

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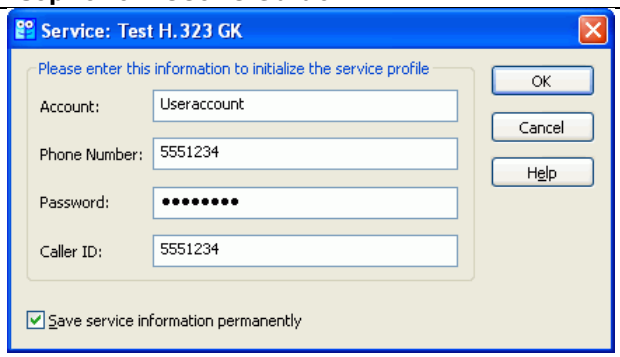

- Enter the required information and click the **OK** button

Select Save service information permanently if you do not want to enter this information every time you select this profile.

5.2 NAT traversal and SIP

Symptoms that you have problems with NAT:

- Audio goes only one way (either inbound or outbound).
- Incoming calls cannot reach you.

Note: If you experience problems with NAT your best friend(s) is your ITSP support or/and network administrator. Always consult them when in doubt.

Most likely you are behind a NAT if you use an Internet gateway connecting your local network to a broadband network via a DSL or cable modem, use Internet Connection Sharing in Windows XP, or are on a private network (your computer has an IP address like 10.x.x.x, 192.168.x.x, 172.16.0.0 - 172.31.255.255).

If your service profile allows you to do this, you may check if **SJphone™** is behind NAT. To do so, switch to **Advanced Mode**.

You are NOT behind NAT if you see the following:

- NAT/Firewall: Open Internet

You are behind NAT if you see the following:

- NAT/Firewall: Full Cone NAT
- NAT/Firewall: Restricted Cone NAT
- NAT/Firewall: Port Restricted Cone NAT
- NAT/Firewall: Symmetric NAT
- NAT/Firewall: NAT of Unknown Type
- NAT/Firewall: Symmetric Firewall (Strictly speaking, a symmetric firewall is not a NAT device but **SJphone™** works behind a symmetric firewall as it were behind NAT.)

It is unclear if you are or are not behind NAT if you see the following:

- NAT/Firewall: Unknown
- NAT/Firewall: Blocked

SJphone™ and NAT

NAT (stands for Network Address Translation) is a widely-used solution that allows multiple PCs on a private network to share a single, globally accessible IP address. A main reason to use NAT is the fact that IP addresses rapidly become scarce. In addition, NAT provides an extra layer for network security by obscuring a private network from an outsider. Please note that NAT is not the same thing as a firewall or proxy server.

Unfortunately, NAT makes communications for **SJphone™** and other Internet phones much more difficult. The situation is even worsened by the fact that there are several types of NAT, each imposing

different restrictions, and different methods are required to traverse a NAT. **SJphone™** provides most of them.

NAT traversal methods

Below are NAT traversal methods that **SJphone™** supports. Which is to be used, depends on your ITSP, on your network configuration, etc. Always consult your ITSP support or/and network administrator when in doubt.

1. Symmetric media streams: **SJphone™** is preconfigured to use it. No additional configuring is required.
2. STUN protocol
3. rport SIP extension
4. OPTIONS request registration refresh
5. Outgoing INVITE transaction refresh

Settings in the Call through SIP Proxy profile type for the NAT traversal

Attention! You must completely understand what you are doing while changing these settings. Always consult your ITSP support or/and network administrator when in doubt.

Consult your ITSP or SIP Proxy administrator on which method to use:

1. A second proxy is used for NAT traversal:

Enter the proxy's DNS name or IP address to the Proxy for NAT field on the **SIP Proxy** tab. Enter the port number if it is not a default value (5060) and there is no this information in the NAPTR/SRV DNS records.

Note: If a second proxy for NAT is used, SIP messages do not include the STUN information.

2. One proxy supports NAT traversal or rport.

Disable STUN usage by clearing the Use discovered addresses in SIP option on the **STUN** tab.

3. One proxy does not support NAT traversal

Select the Use discovered addresses in SIP option on the **STUN** tab. Enter STUN settings on the STUN tab. For the most cases the default settings are correct.

This is the worst case and you may experience problems with calls. Consult your ITSP (Internet Telephony Service Provider) support or/and network administrator in this case.

5.3. Compression and Advanced Audio Settings

(available in the **Advanced Mode** interface only and may be unavailable for some service profiles)

To change the compression or advanced audio settings, click the **Compression** or **Advanced Settings** buttons on the **Audio** tab.

Attention! You must completely understand what you are doing while changing these settings or you can severely degrade **SJphone™** performance.

Compression settings

Codec preferences

When two VoIP systems are establishing a call, they negotiate an audio compression codec they are going to use. Which codec to choose depends on many factors: what codecs are installed on the systems, bandwidth limitations, desired sound quality, etc. You may adjust **SJphone™** codec preferences on the **Compression Settings** panel. During the negotiation, **SJphone™** offers to the remote system the first codec from the list. If the remote system rejects the codec, **SJphone™** offers the next one from top to bottom until they both accept the codec. You may arrange the codecs in the desired order using the **Move Up** and **Move Down** buttons. The **Use Default** button selects the default codec preference.

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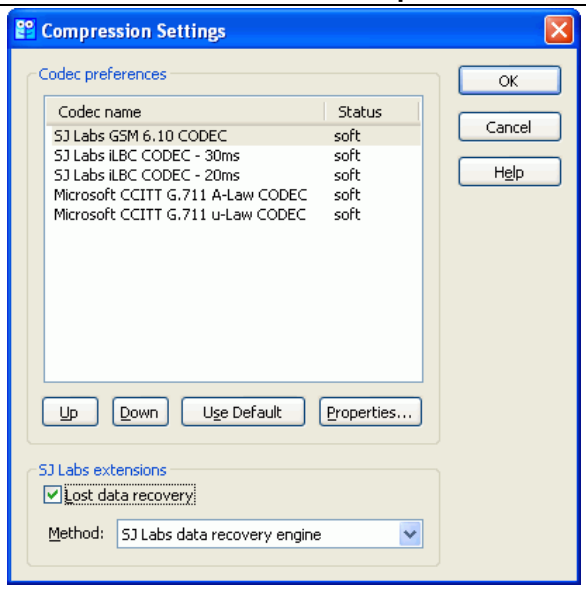
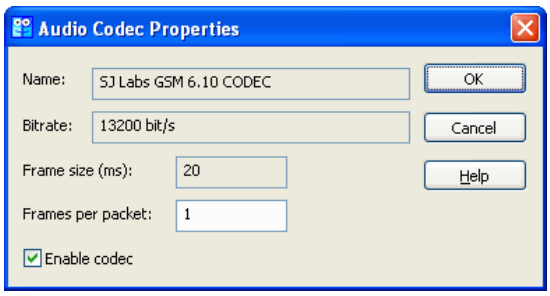




Codec preferences

Codec name	Shows codec name
Status	Shows codec status. May be either: • Soft: the codec is implemented as a software driver • Hard: the codec is implemented as a hardware device • Disabled: the codec is disabled. A codec may be enabled or disabled on the **Audio Codec Properties** panel.

Buttons

Up	Click this button to move the selected codec up
Down	Click this button to move the selected codec down
Use Default	Click this button to use a default codec preference
Properties	Click this button to activate the **Audio Codec Properties** panel

SJ Labs Extensions

Lost data recovery	Check this box to enable the advanced data recovery system
Method	Select the method of the advance data recovery from the list of available methods

If you select a codec and click the **Properties** button, or double-click the codec, its **Audio Codec Properties** message will appear: This panel shows some of the codec's basic properties. You may also enable or disable the codec on this message.

The current **SJphone™** version supports the following codecs:

- G711 A –Law and G711 u –Law codecs. They provide good sound quality but low voice compression. Both codecs require 64 kbit/sec bandwidth and may be recommended for wide bandwidth networks, such as LANs.
- GSM and G.729 codecs may be recommended if network bandwidth is limited, for example for long-distance calls or for dial-up connections. G729 codec provides good voice compression at 8 kbit/sec plus overhead. GSM codec provides an average sound quality and good voice compression at 13 kbit/sec plus overhead.
- iLBC codec may be recommended for networks with a large number of lost packets. It provides good sound quality at 13.3 kbit/sec plus overhead.

Connection	**Codecs**
Dial-up modem	iLBC 20 msec (possibly)
	iLBC 30 msec (with possible small congestions)
	GSM (with possible small congestions)
	G.729
ADSL 64 kb/sec	G.711 (with possible small congestions)
	ILBC
	GSM
	G.729

Wideband (128 kb/sec and wider)	All codecs

Note: if your connection is asymmetric (ADSL 128/64, for example), codecs should match the narrowest channel. Also, any other data transfer activity like downloading files or web browsing on a low-bandwidth connection may sharply reduce the bandwidth available for **SJphone™**.

SJ Labs Extensions

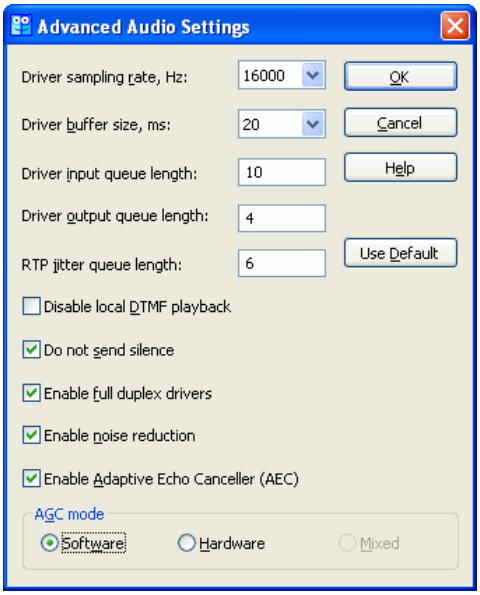
Lost data recovery allows you to obtain good sound quality over poor Internet connections. The current **SJphone™** version supports SJ Labs data recovery engine.

Attention! Use it only while communicating with other **SJphone™**-compatible software.

SJ Labs data recovery engine sends additional IP packets. This doubles bandwidth requirements but completely restores up to 50% of lost packets. This feature is useful for wideband but unstable connections such as Mobile Internet, WaveLan, or Radio Ethernet.

Advanced Audio Settings

Click the **Advanced Settings**... button on the **Audio** tab to reach the **Advanced Audio Settings** panel.



If **SJphone™** plays back DTMF signals too loudly, you may disable local DTMF playback by selecting the Disable local DTMF playback option.

Do not Send Silence: If this option is selected, **SJphone™** does not send IP packets when you are silent, or you mute **SJphone™**. That reduces network load.

The following settings allow you to change sizes of various internal **SJphone™** sound buffers and queues if you experience **problems with sound**.

Do not change them unless you completely understand what you are doing, or you may severely degrade the **SJphone™** performance.

Driver sampling rate	You may select which sampling rate is to use. **SJphone™** will automatically resample the sound if that value does not match the sampling rate of a used codec
Driver buffer size:	Sets the driver buffer size in msec. Please note, that the value of this setting was in bytes in previous **SJphone™** versions
Driver input queue length	Sets the number of buffers used in input and output driver queues.
Driver output queue length	
RTP jitter queue length	Sets the number of packets **SJphone™** used to compensate for the network jitter.

The following settings may be enabled only if the Enable Direct Sound for improved audio performance option is enabled on the **Audio** tab.

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Enable full-duplex drivers	When enabled, makes the input and output sampling rates equal, which is necessary for Automatic Echo Canceller to operate stably
Enable noise reduction	When enabled, reduces sound noise. Active only when Enable full-duplex drivers is enabled.
Enable Adaptive Echo Canceller (AEC)	Enable this feature to reduce echo that your respondents may hear.
AGC mode	You may select which type of microphone automatic gain control **SJphone™** uses. Effective only when Automatically adjust microphone volume is enabled on the **Audio** tab Software: **SJphone™** adjusts sound level itself without changing system sound settings. Use this option when microphone sensitivity is poor. Hardware: **SJphone™** adjusts sound level itself by changing system sound settings.

5.4. Command Line Switches

You can run **SJphone™** from a command line.

In order to do this, click the `Start->Run->Browse` to find the place where `SJphone.exe` is installed, or run it from the Windows Command Prompt: `C:\Program Files\SJphone 1.65\SJphone.exe`

Syntax: `SJphone [<DialString>][<keys>]`

where:

`<DialString>` is either an Extended H323 Address or SIP URI.

`<keys>` is one of the following:

`/?` or `/h` or `/help` invokes command line switch help.

`/lf <FileName>` - **SJphone™** writes its log to `FileName`. This key may be used with other keys.

`/bugreport` starts writing a bug report immediately after **SJphone™** startup.

`/lf <FileName>` **SJphone™** writes its log to `FileName`. This key may be used with other keys.

`/d` - **SJphone™** hangs up all existing calls.

`/callto <URL>` - **SJphone™** calls to the specified `URL`, where `URL` is an URL specified as:

a host name (example: `jd.example.com`)

an IP address `+type=ip` (example: `192.168.1.12+type=ip`)

a telephone number`+type=phone` (example: `1357999+type=phone`) - a call through a default Gateway or Gatekeeper, whichever is selected by the default profile.

a telephone number`+gateway=<gatewayname>+type=phone` (example: `1357999+gateway=h323gateway.example.com+type=phone`) - a call through a specified Gateway

a SIP URI`+type=sip` (example: `jd@example.com+type=sip`) - a call to the user `jd` at the host `example.com` using the SIP protocol.

any string`+type=hangup` (example: `hangup+type=hangup`) - all active calls will be terminated

`<DialString>+type=dialstring` (example: `h323:gk:h323id:Gedy+type=dialstring`) -a call to user `Gedy` registered with the current gatekeeper.

The parameter `existingcalls` may be added to any `callto` string. This parameter specifies what to do with active calls.

`+existingcalls=leave` (default) leaves all calls in their current state.

`+existingcalls=hangup` terminates all active calls before a new call.

`/postpone <dialstring>` **SJphone™** dials the `dialstring` using its current profile. The `dialstring` should be valid for the profile.

`Callto` example: `callto:h323:gk:h323id:Boss+type=dialstring+existingcalls=hangup -` all active calls will be terminated and **SJphone™** will start a call to user `Boss` registered with a current **SJphone™** gatekeeper.

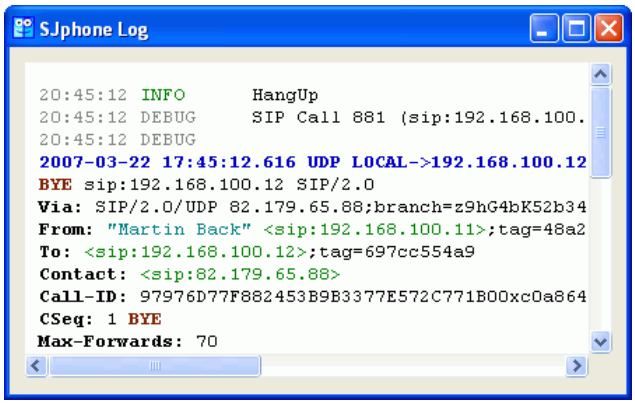
5.5. Log Window and Multimedia Engine Statistics

(available in the **Advanced Mode** interface only and may be unavailable for some service profiles)

Log Window

The log window shows you technical information about **SJphone™** operation and call progress.

To understand this information, you need to be familiar in depth with the SIP or ITU-T H.323 protocol details.



To enable the SJphone™ Log window,

- Right-click the **SJphone™** system tray icon or skin and select **Show Log** on the shortcut menu.

The **Log** window will appear.

To disable the SJphone™ Log window,

- right-click the **SJphone™** system tray icon or skin and clear **Show Log**.

Alternatively, you can start **SJphone™** with the following key:

`SJphone.exe /lf <Filename>`

where

`<Filename>` is log output to the file `Filename`.

SJphone™ will write its log to the specified log file.

Multimedia Engine Statistics

You may also see various multimedia statistics. Right-click the **SJphone™** system tray icon or skin and select **MME Statistics**. The **MME statistics** window will appear.

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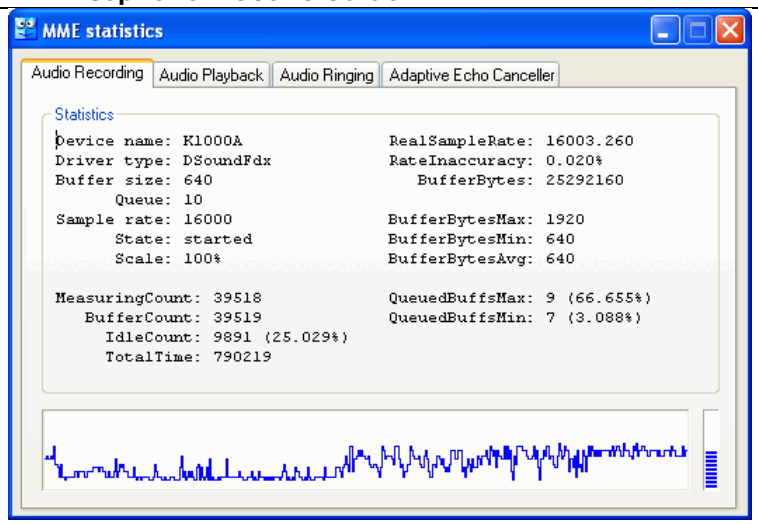
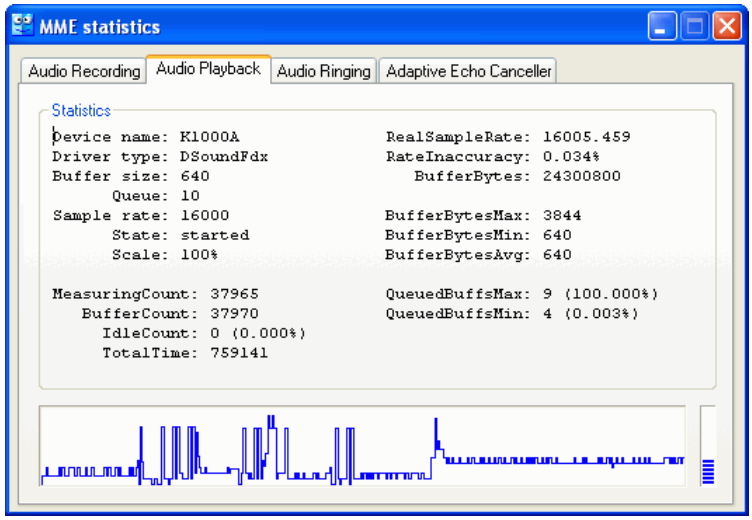
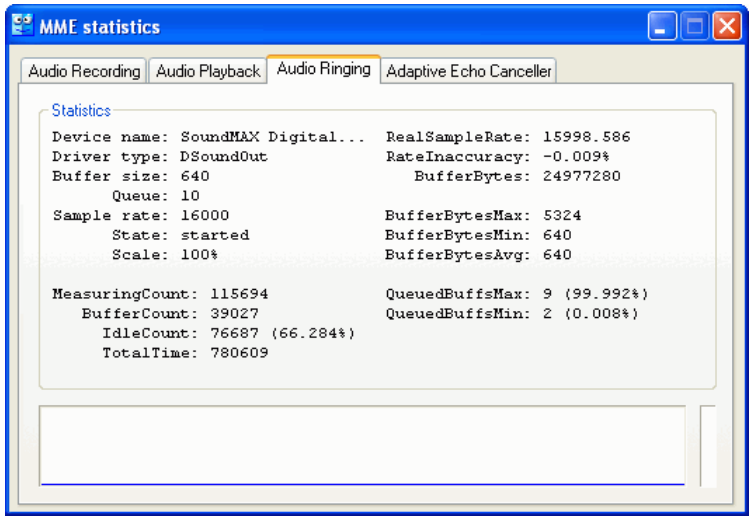
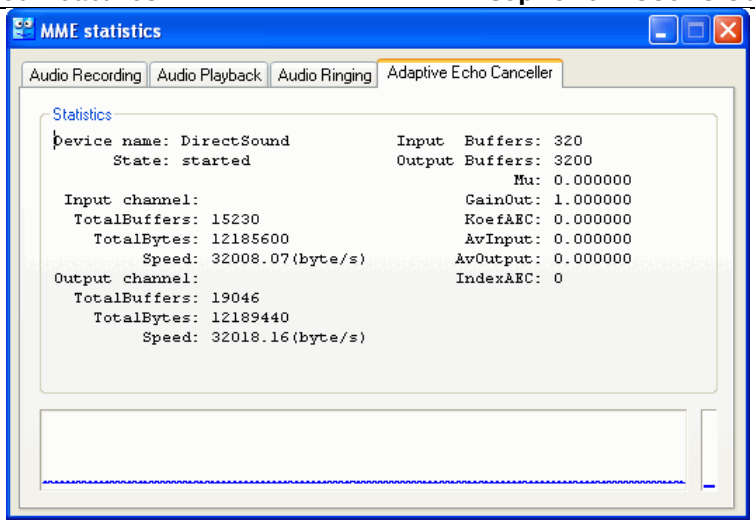

MME statistics
Audio Recording
Audio Playback
Audio Ringing
Adaptive Echo Canceller
Statistics
Device name: K1000A
Driver type: DSoundFdx
Buffer size: 640
Queue: 10
Sample rate: 16000
State: started
Scale: 100%
MeasuringCount: 39518
BufferCount: 39519
IdleCount: 9891 (25.029%)
TotalTime: 790219
RealSampleRate: 16003.260
RateInaccuracy: 0.020%
BufferBytes: 25292160
BufferBytesMax: 1920
BufferBytesMin: 640
BufferBytesAvg: 640
QueuedBuffsMax: 9 (66.655%)
QueuedBuffsMin: 7 (3.088%)

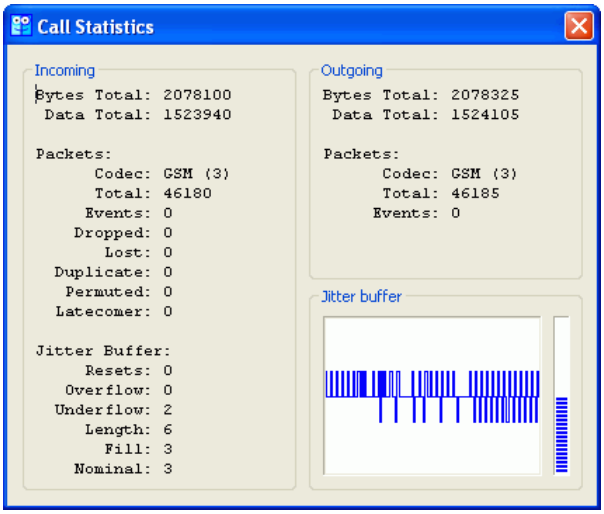

MME statistics
Audio Recording
Audio Playback
Audio Ringing
Adaptive Echo Canceller
Statistics
Device name: K1000A
Driver type: DSoundFdx
Buffer size: 640
Queue: 10
Sample rate: 16000
State: started
Scale: 100%
MeasuringCount: 37965
BufferCount: 37970
IdleCount: 0 (0.000%)
TotalTime: 759141
RealSampleRate: 16005.459
RateInaccuracy: 0.034%
BufferBytes: 24300800
BufferBytesMax: 3844
BufferBytesMin: 640
BufferBytesAvg: 640
QueuedBuffsMax: 9 (100.000%)
QueuedBuffsMin: 4 (0.003%)


MME statistics
Audio Recording
Audio Playback
Audio Ringing
Adaptive Echo Canceller
Statistics
Device name: SoundMAX Digital...
Driver type: DSoundOut
Buffer size: 640
Queue: 10
Sample rate: 16000
State: started
Scale: 100%
MeasuringCount: 115694
BufferCount: 39027
IdleCount: 76687 (66.284%)
TotalTime: 780609
RealSampleRate: 15998.586
RateInaccuracy: -0.009%
BufferBytes: 24977280
BufferBytesMax: 5324
BufferBytesMin: 640
BufferBytesAvg: 640
QueuedBuffsMax: 9 (99.992%)
QueuedBuffsMin: 2 (0.008%)



Call Statistics

You may also see various multimedia statistics for each call. Right-click the call and select **Call Statistics**. The **Call Statistics** window will appear



5.6. H.323 Address Syntax

An extended H.323 address can be represented as the following string:

```
h323:<AddressType>:<Address1>[:<Address2>]
      where <AddressType> = ip
                <Address1> = <ip address> | <dns name>
            <Address type> = gw
                <Address1> = <GW ip address> | <GW dns name>
                <Address2> = <phone number to dial>
            <Address type> = gwhunt
                <Address1> = <phone number to dial>
                <Address2> = <File with GW profile>
            <AddressType> = gk
                a)<Address1> = ip
                  <Address2> = <ip address> | <dns name>
                b)<Address1> = h323id
                  <Address2> = <registered user alias with GK>
                c)<Address1> = e164
                  <Address2> = <user's phone number registered with GK or
                  respondent's phone number in PSTN>
```

All parameters are not case-sensitive. You may use "+", "(", ")", "-", and space in telephone numbers.

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H.323 Address Examples

If you want to call

- A host `jd.example.com` (`192.168.1.12`)
  Type in the **Call To:** field: `H323:IP:jd.example.com` or `H323:IP:192.168.1.12`
- A phone number `135-7975` through the H.323 Gateway `h323gateway.example.com`
  Type in the **Call To:** field: `H323:GW:h323gateway.example.com:1357975`
- A user with the nickname `doe` registered with the H.323 Gatekeeper
  Type in the **Call To:** field: `H323:GK:H323ID:doe`
- A phone number `135` registered with the H.323 Gatekeeper
  Type in the **Call To:** field: `H323:GK:E164:135`
- An external phone number `9-5557975` through the H.323 Gatekeeper
  Type in the **Call To:** field: `H323:GK:E164:95557975`

5.7. SIP URI Syntax

A SIP URI can be represented as the following string:

```
sip:[<user [:password] | phone-number @>]<DNSname |
     ipaddress>[;<parameter1=paramvalue1>[;parameter2=paramvalue2]...]
     where <parameter1> = user
           <paramvalue1> = <ip> | <phone>
           <parameter2> = transport
           <paramvalue2> = <tcp> | <udp>
```

All parameters are case-sensitive.

Comprehensive SIP URI syntax is described in RFC3261.

SIP URI Examples

If you want to call

- A host `jd.example.com` (`192.168.1.12`)
  Type in the **Call To:** field: `sip:jd.bigcom.com` or `sip:192.168.1.12`
- A user `johndoe` at `jd.example.com`
  Type in the **Call To:** field: `sip:johndoe@jd.example.com`
- A phone number `555-7975` through a SIP gateway `sipgateway.example.com`
  Type in the **Call To:** field: `sip:555-7975@sipgateway.example.com;user=phone`

5.8. DTMF Signal Sending

Depending on the VoIP protocol and service, DTMF signals may be sent using various methods. Which method is to choose depends on a selected service. The method is specified in the service profile.

SIP VoIP Protocol

SJphone™ supports 3 methods:

1. In-band: DTMF signals are sent as digitized sound in the RTP channel
2. RFC2833: DTMF signals are sent as special packets controlled in the RTP channel
3. SIP INFO: DTMF signals are sent in the `INFO` message in the SIP session control channel

If the SIP INFO method is selected, the DTMF signals are always sent in the INFO message. If the RFC2833 method is selected and the remote system does not support this method, the In-band method will be used.

For a direct call, the remote user does not hear DTMF signals when the SIP INFO or RFC2833 methods are used. For a call through a SIP proxy, this depends on the proxy. Some convert DTMF signals into sound, some do not.

H.323 VoIP Protocol

SJphone™ supports 4 methods:

1. In-band: DTMF signals are sent as digitized sound. This is the choice if you want to control an external telephone device such as a fax machine, answering machine, etc.
2. Q.931 keypad: DTMF signals are sent through the Q.931 KeypadFacility Information Element of Q.931 messages.
3. H.245 alphanumeric: DTMF signals are sent as one of the out-of-band method in H.245 protocol messages.
4. H.245 signal: DTMF signals are sent as one of the out-of-band method in H.245 protocol messages.

The remote user will hear DTMF signals when the In-band method is used.

5.9. Used Ports

SJphone™ uses the following ports:

Port	Protocol	Description
1719	UDP	H.323 Gatekeeper RAS port
1720	TCP	H.323 Call Signaling
1812	UDP	RADIUS server in **SJphone™**
3478	UDP	STUN service
3479	UDP	STUN service
5002	TCP	MLP protocol server
5003	UDP	Neighborhood service
5060	UDP	SIP UAS
5060	TCP	SIP UAS
5061	TCP	SIP UAS (TLS)
5222	TCP	Jabber
5223	TCP	Jabber (TLS)
49152-65535	UDP	RTP, RTCP multimedia streaming

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VI. Troubleshooting

Sound quality can vary significantly depending on your sound card, microphone, speakers, and Internet connection. Make sure that you have a good microphone and speakers, and the latest driver for the soundcard.

Check your CPU usage. It should be less than 50%, otherwise you might consider upgrading your CPU.

If you experience a bad sound quality, turn off the Automatic Gain Control feature on the Windows sound control panel. Go to **Control Panel**, then to **Sounds, Speech, and Audio Devices**, then to **Sounds and Audio Devices**, then to **Volume**, and to **Advanced** for **Microphone**.

6.1. Start-up Troubles

If **SJphone™** does not start up properly, try the following:

- Check that other audio recording or playing software is not running.
- Check that other H.323 or SIP software, such as NetMeeting or Windows Messenger, is not running.

If the problem persists, contact SJ Labs technical support.

6.2. Sound Issues

If you experience bad sound quality, clicks, long sound delay, or dropped sound fragments, you may try to adjust **Advanced Audio Settings**.

Do not change them unless you completely understand what you are doing, or you may severely degrade the **SJphone™** performance. You may always return to the default values by clicking the Use Default.

Audio channel latency time

If you want to decrease channel latency time, decrease either Driver buffer size, Driver output queue length, RTP jitter queue length, or all of those parameters. Please, note that Driver input queue length does not change the latency time.

Recommended Driver output queue length is 2 if DirectX is installed and used, and 4 without DirectX. You may check DirectX status on the **Audio** tab.

Distorted sound and clicks

If you hear clicks or sound is distorted:

- on the remote side, increase Driver input queue length.
- on your side: increase Driver output queue length. If this does not help: increase Driver buffer size.

Sound drops

If you experience sound drops,

- Check the number of lost and out-of-order packets on the **Main** panel (without skins). Go to the **Connection Issues** section for details.
- If selected, clear Automatically adjust silence detection level on the **Audio** tab.
- Try to increase RTP jitter queue length.

Sometimes, when **SJphone™** is connected with a regular telephone through a Gateway, direct and inverse sound lags may be different, because audio packet buffers may have different size in **SJphone™** and the Gateway.

If the problem persists, contact SJ Labs technical support.

6.3 Sound Card Issues

You can adjust the **SJphone™** sound characteristics using standard Windows audio controls or controls on the Main panel. The **Mic** and **Spk** controls are connected with the standard Windows audio controls. Any changes in their settings automatically change the corresponding settings at the Windows volume control panel.

You may also use an **Audio Wizard** to fine tune your sound system for work with **SJphone™**.

SJphone™ requires a full-duplex sound card. You can check whether your sound card is full duplex. Try to record sound from the microphone with the Sound Recorder while another copy of the Sound Recorder plays back any sound file. If you can do this, your sound card is full duplex. If you cannot, your sound card driver may be not full duplex. Obtain a full duplex driver from the sound card manufacturer.

You can use the automatic microphone gain feature in your sound card to automatically adjust the input sound level. Consult your sound card documentation for details. Some sound cards do not have the automatic microphone gain feature. You can select the **Automatically adjust microphone volume** box on the **Audio** tab to let **SJphone™** automatically adjust the microphone sound level.

If there is a strong background sound when the Automatically adjust silence detection level on the **Audio** tab is activated, **SJphone™** may not detect silence correctly, and fragments of your speech may be lost. In this case, turn this feature off.

If your sound card supports DirectSound, select the Enable Direct Sound for improved audio performance box on the **Audio** tab. This will decrease sound latency time. Most sound cards from Turtle Beach, Yamaha, Creative Labs (excluding the Ensoniq-based AudioPCI types), Diamond, Crystal, or Microsoft USB speakers support this feature.

If you are using DirectX option, make sure that you have DirectX 9.0+ installed. You can check it with `dxdiag.exe`.

If you hear hissing noise from your speakers while talking, try the following:

- Move the microphone farther from the speakers.
- Decrease the microphone or speaker sound level.

If the problem persists, contact SJ Labs technical support.

6.4. Connection Issues

If your host or the host being called is on a private network, VoIP communication using the H.323 protocol is possible only through a Gatekeeper. This is a feature of the H.323 protocol.

If you experience problems in connecting with a remote host, try to check the connection using the `Ping` command in the DOS prompt. Refer to Windows documentation for details.

The `Ping` command can also be used to check communication delay with the host being called. Reasonable roundtrip time should be 100-200 ms for a dedicated and 200-300 ms for dial-up connection. The delay should vary within 10% of average connection delay. If the delays are unexpectedly long, contact your Internet provider or system administrator.

Lost packets should not exceed 10-15% for good communication quality. If you loose speech fragments due to a large number of lost packets, try to adjust codecs or turn on the **SJ Labs data recovery engine**. Go to the **Compression and Advanced Audio settings** section for details, or contact your Internet provider or system administrator for assistance.

If the number of out-of-order packets is large, your network is overloaded or unstable. Congestion in packet switched networks can cause packets to take different routes to reach the same destination. Packets may arrive in different order resulting in garbled speech. To prevent this, **SJphone™** stores received packets in a special buffer and reorders them. That is why even large number of out-of-order

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packets does not always deteriorate sound quality if the packets arrive within a certain time. If a packet comes too late, it gets lost.

If the problem persists, contact SJ Labs technical support.

6.5. NAT Issues

Symptoms that you have problems with NAT:

- Audio goes only one way (either inbound or outbound).
- Incoming calls cannot reach you.

Read the **NAT traversal and SIP** section for details.

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